SOME ASPECTS OF A PACKET SWITCHED COMPUTER COMMUNICATION SYSTEMS

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CERTIFICATE

Certified that this work on "SOME ASPECTS OF A PACKET SWITCHED COMPUTER COMMUNICATION SYSTEM" has been carried out under my supervision and that this has not been submitted elsewhere for a degree.

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ABSTRACT

Some aspects of a packet switched computer communication system have been considered. While circuit switching is apt for voice communication using telephone networks, computer communication needs packet switching. After explaining the concept and need for packet switching, the factors associated with its implementation have been briefly dealt with.

The topological design optimization of a computer communication networks is influenced by message delays, reliability and path lengths. All these factors have been studied in detail. The waiting time distribution of packets moving through intermediate nodes enroute their destination is based on queueing system model. Graphs are plotted, showing message delays for various traffic intensities and different packet lengths.

Reliability means the availability of a system for communication between two nodes. Various reliability criteria and reliability measures have been mentioned. Reliability measure based on minimal cut sets has been described which saves a lot of computation and gives a good lower bound on system reliability.

Transmission delay is a function of average path length of the network. Graph theoretic considerations have been used to provide a simple criterion for finding the average path length of a network.

The techniques used to transmit packets over the existing telephone links and the effects of various channel impairments on the transmitted data have been briefly described.

A typical computer communication system for defence applications has been described using the above concepts.

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CHAPTER 1

INTRODUCTION

Digital computers came on the scene of the technologic world after the Second World War and though the principles of digital communication were well known during the Second World War and theoretic developments continued during the subseque period, digital communication in practice could not be realised effectively because of the hardware limitations. In the last two decades or so with the advancement of digital hardware technology digital communication has come to stay because of its many attractive features outlined as below.

1. The relative simplicity of digital circuit design and the ease with which one can apply integrated circuit technique to digital circuitory.
2. The ever-increasing use and availability of digital processing techniques.
3. The widespread use of computers in handling all kinds of data.
4. The ability of digital signals to be coded to minimise the effects of noise and interference.

5. The frequent regeneration which eliminates the effect of noise and distortion.

6. The size, weight and the cost of equipments are reducing with improvement in digital technologies.

7. Digital communication gives a better alternative to use the total channel capacity.

During this period when digital communication was perfected by communication technologists as a means for effective communication, digital computer had increased in their size, the data handling capabilities and sophistication. Naturally one day the two technologies of computer and communication had to merge and this resulted in what is now called as computer communication. For computer communication one has to develop networks so that communication from a machi to another machine or machine to a man or vice versa can be established. The basic idea is that using the existing communication links one could design ways and means so that communication between machines can be carried out with a high degree of reliability and fidelity.

Towards this end, work started as early as late 60's and now computer network is operational over many communication links for commercial purposes in the United States and other developed countries. In our country also,

impairmental links have been established for computer communication purposes. The communication medium to link two modes serving machines is the usual communication channel, for example, the usual telephone line or radio network or a satellite link. A satellite link obviously gives the best channel characteristics because many impairments that exist in usual communication channels do not exist on a satellite link. Communication between machines is basically transfer of digital data from one to the other. This can be carried out when digital data is continuously transmitted, as in message switching or the data can be segmented in packets and transmitted, as in pocket switching. The first one is analogous to voice communication whereas the second one is specific to computer communication. The computers at nodes may also include a set of terminals. Thus, objective of computer communication network would be to provide communication between a set of computers so that the resonances can be mutually shared by computers located at distant ends. It is not economically feasible to provide large computing facilities at all the places. Generally, a large computer system could be available only at selected places like Regional Computer Centres, whereas local small computers or

terminals might be available at many varied geographical destinations.

It is then imperative to establish contacts between the computers so that resources at one location can be utilised by other machines. The resources which might be shared would include software, data bank and storage facilities. Resource sharing might be on time sharing basis or in any other fashion.

Before we really go into the description of packet-switching one should realise that we are transferring data on a communication link between one place to the other. From communication point of view the data must be secure and unambiguous when it is transmitted from one point to the other i.e. the transmission link should not degrade the data.

For this particular purpose one will have to use a communication network in such a manner that the degradation is rather minimal and if a packet or block of bits is not received correctly then we should cater for its correction and retransmission. This actually brings us to the problem of how one can transmit digital data over the communication link so that it is accurately received at the destination. The impairments because of a link, whether it is terrest-erial or a radio, or a satellite, are because of dispersion,

intersymbol interference, thermal noise and so on.

Thus in a nutshell, the objective is to provide a means to transmit data from one point to the other at a fidelity rate which is sufficient for computer operations, and at a cost which is economically viable. This also would mean that if because of any reason a particular path is faulty (it may be down, the noise may be excessive etc.) alternate path would be available like in present day telephone routing, for packets from one computer to another, the capacity of the system is utilised dynamically, a means by which resources at one centre is shared by others.

The field of computer communications has grown rapidly in its applications during the past decade. The concept of resource sharing has added to new dimensions to the facilities for the user. With this concept the user can make use of the capabilities of all the interlinked computers and terminals. Computer networks are derived from a combination of computers and telecommunications. Organisational complexity in computers shot ahead with the concepts of stored program and of programming languages while solid state electronics set the pace for a rapid improvement in the speed and cost of digital processing. This process is still continuing.

We now have a technological convergence of computers and communications, both sharing the same kind of logic, storage, switching and transmission. The basic technical problems to cater for in a computer communication system are: to economically and reliably transmit packets from one location to another, to handle interface and interprocess communication among the computers, terminals and communication equipment, to control information flow and to predict system behaviour, to efficiently share software and hardware resources. Some more have been listed in Chapter 2. All this informations has been scattered in various literatures.

The factors affecting the topological designs of a computer communication network have been studied in detail along with the concept and need for packet switching and the communication problems while transmitting packets from one node to the other.

While establishing the need for packet switching in computer communication systems, the other switching techniques and the relative advantages /disadvantages have been described in Chapter 2. Various design consideration like routing, flow control, protocols, topological optimisation etc. and the hardware problems in implementing

packet switching have been briefly mentioned. Some definitions associated with a packet switched system have also been given.

The topological design considerations include the message or packet delays, the reliability and the average path length of the network. Each of the above factors have been studied in detail. For arriving at the expression of message delays the message length is considered to be random and not fixed. Also, messages arrive at random at a node. The delays for fixed message lengths can be derived from this expression. These delays are basically the queueing delays and the analysis of waiting time distribution is based on queueing theory. These are covered in Chapter 3.

By reliability we mean the availability of a communication link for packet transmission. Intensive research is being carried out to design maximally reliable computer networks under a set of given constraints. There are numerous ways of defining reliability and numerous ways of calculating it. One such method using cut sets has been described in Chapter 4 which saves a lot of computations and gives a good approximation to the system reliability.

Transmission delay in a network is a measure of path lengths between various nodes. Though transmission delay can be reduced using various techniques, our emphasis is on optimising the path lengths and correlating it with number of links which provide the criterion for transmission costs. We discuss these in Chapter 5.

It is of interest to know the communication problems involved in transmission of packets over a link in a packet switched computer communication system. The modulation, techniques, need for coding and various channel impairments have been described in Chapter 6. As mentioned earlier, the world's telecommunication network is still basically a telephone network. The same channel is used for computer communication with the addition of modems and different type of switching e.g. packet switching. Various modulation techniques have been compared. Brief description has been made of the equalizers used to minimise the inter-symbol interference.

We try to assimilate all the studies by taking a typical case study of a computer communication network for defence application in Chapter 7 and giving a system design. Finally we conclude the thesis by reviewing and pointing out the topics for further study.

CHAPTER 2

PACKET SWITCHING

There are mainly three ways of establishing a computer communication network and that is by; [2]

1. Circuit switching
2. Packet switching
3. Message switching

In the following sections we will briefly elaborate on the three types of switched networks and show that the packet switched networks are best suited for the present day computer communications.

2.1 Circuit Switching:

We are quite familiar with circuit switching in the telephone networks. It can be applied in the same way for the computer communication networks. Establishing a call through a circuit switched network requires a path to be found, along which circuits are available that can be connected end to end to make the required path. That is, a physical path must exist from the source to the destination. This is accomplished by the control circuitory. The stream of data bits emerging from a source reaches the other end with the same time pattern delayed by a constant

interval, depending on circuit parameters. The circuit switched network has a constant bit rate depending on the bit rate of a segment of the path having the least capacity.

The three main component of a circuit switched network are switches, controllers and network for control system, as shown in Fig. 2.1.

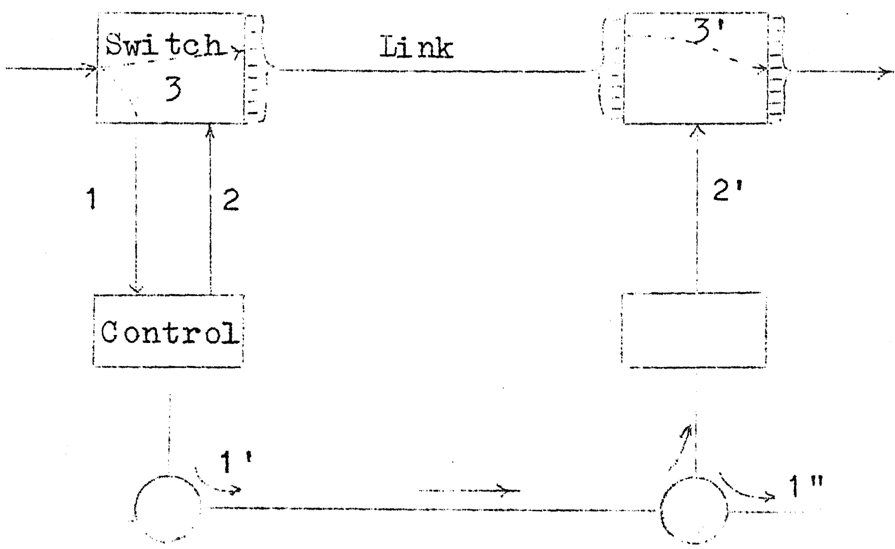


Fig. 2.1: Schematic diagram of a circuit switched network.

In the circuit switched network the source and the destination are connected by a dedicated communication path that is established before the message is actually transmitted and broken at the end. Circuit switching systems are better

equipped to maintain a time frame for users that requires continuity in transmission (like speech). Since the communication path remains tied up for the duration of conversation or data transmission the output signal appears to be a time translation of the input signal. For most voice conversations the analogue voice channel is used in a fashion that is reasonably efficient to the average number of users.

In traditional design of telephone networks the control signals are carried over the same path as voice signals but modern designs, intended for fast call set up and release, permit separate control signals in their own store and forward network.

2.2 Packet Switching [1 , 3 , 4]

Over voice communication network using telephone channels,circuit switching is very appropriate, because of uniform telephone bandwidth of 4 KHz and the duration of calls which justifies the cost of establishing the switched circuit. However data traffic in computer communications does not fit in because we have a wide variety of data signalling rates. The information handled between computers and terminals or vice versa or between computers themselves is not primarily of synchronous nature and is better

regarded as a sequence of messages or data blocks. The data flow may not be continuous, unlike the case in the circuit switched systems. This is the reason for the development of packet switching. In packet switched networks short messages called packets are handeled individually and separately by the network. In its simplest form packet si switching consists merely of directing packets on their way to their destination through various nodes having store and forward capabilities. Depending on the capacity, data packets can also be multiplexed by interleaving packets from various data streams. Because packets are stored at each intermediate node which switches them and forwards them to the next node on their way to the destination, this is also known as store and forward method. Packets can be entered into or removed fr the network at the speed which suits the terminals and the computers. Therefore the packet switching network acts as a speed changer also.

An important aspect of packet switching is to provide an efficient and reliable computer communication system in which computer resources of each one of them, such as programmes, data, storage, special purpose software and hardware etc. could be shared, amongst the computers and terminals. Further, the packet switching techniques applied

in a resource sharing system have helped bring down the cost of the system while increasing its reliability, responsiveness and capacity.

Another feature of packet switching is its interfacing capability with a multi-accessed computer. By multi-access we mean that the computer can engage in more than one conversation with other computers or terminals at the same time.

The switching nodes are referred to as Interface Message Processors (IMPs). An IMP accepts traffic from a computer attached to it, formats it into packets and routes it towards its destination. In case the onward transmission is not feasible due to channel being busy, the packets join the queue at that node.

Each IMP in the network, on receiving a packet examines the header containing the destination, makes a routing decision and passes it on towards its destination possibly through several intermediate IMPs. Thus a packet proceeds from IMP to IMP or node to node in making its way to its destination. The destination IMP collects the packets, reformats them into messages in the proper sequence and passes them onto the destination computer terminal. By means of both hardware and software based error control

techniques, each IMP checks the correctness of each packet. If any packet is received incorrectly the IMP does not acknowledge its receipt and the preceeding IMP retransmits the packet, may be on a different path.

A packet switching system is faced with two major problems viz. routing and flow control. Routing is to route the packets to their destination and flow control is to avoid congestion at any particular node.

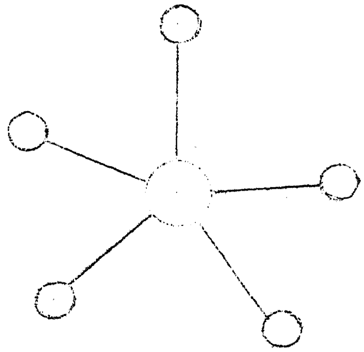
2.3 Message Switching [1]

The packet switching is an offshoot of message switching where a system accepts, transmits and delivers discrete entities called messages unlike continuous information like in voice communication. Like packet switched systems in this system also, no physical path is set up between the source and the destination. No resources viz. capacity, buffer storage etc. are allocated to its transmission in advance at the destination. Rather, the source includes a destination address at the beginning of each message. The message switching system uses this address to route the message through the network to its destination, provides error control and notifies its sender its receipt from intermediate nodes. Message switched systems were evolved with the emphasis on reliable delivery of each message, even if it was at

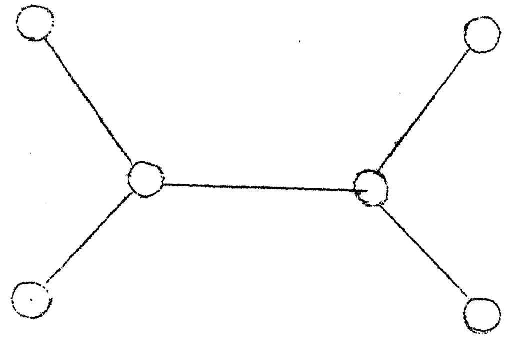
the cost of added delay. Initially this technique was used in handling telegraph messages. Because of the responsibility of the message switching system for safe delivery of messages, each part of the system is subject to checking and provides for recovery from faults. The concept of packet was introduced to keep the delays under control. Besides the reliability of switching nodes the reliability of the network is equally important.

2.4 Network Configurations:

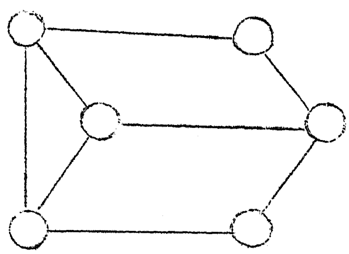
There are many ways to connect nodes and links (channels) to form a computer communication network. This is usually called the topology of the network. Though the reliability criterian and the factors affecting a network design including topology are discussed in Chapters 4 and 5. We shall discuss here some of the common network configurations. Fig. 2.2 shows four ways in which say six nodes can be connected. The first form shown is the star in which there is one central computer and all other computers or terminals are linked to it. Such a network is centred around one computer and the failure of the central node will disrupt the system completely. However, failure of any other node or link does not affect the communication between the other nodes. Therefore the system can be made very reliable



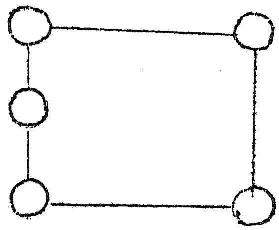
(a) Star



(b) Tree



(c) Mesh



(d) Ring

(●)

Fig. 2.2 Examples of network configurations

(failures of nodes and links disrupting a communication path) by concentrating on the reliability of the central node alone.

Star configuration is a particular form of tree configuration. A tree requires the minimum number of links to connect all the nodes. The links do not merge at one particular node like a star but at many intermediate nodes. The failure of any intermediate will disrupt the communication

between one or more nodes, Since there is only one communication path between any two points. The tree configuration is the most economical configuration and can be used where cost factor is of paramount importance. However the communication path between any two points is not dependant on any particular node or link failure like the central node in the star configuration.

The third example is the mesh configuration and this is the form most often used in most computer communication networks. This configuration provides an alternate path for communication between two points and hence the reliability of the communication between two points increases. But the number of links increase thereby increasing the cost and the path lengths. The maximum number of links is in the fully connected mesh,(a modified Fig. 2.2(c))where all nodes are linked with each other. This type of configuration, though most reliable is not at all cost effective. The network topologies are designed keeping the extent of reliability of communication between two nodes, the cost constraints, path lengths and the utility of various nodes (size and capability of a computer) in mind. These factors have been discussed in the succeeding chapters.

The minimum connected mesh configuration is the ring configuration. It is less economical than the tree configuration but it proves better than the star network. The data can be routed either way round the path i.e. it provides two possible communication paths between any two points in the system. It is therefore a compromise between a tree configuration and a mesh configuration in every respect.

2.5 Advantages of Packet Switching

In the evolution of data communication the initial study of time sharing techniques had led to an assumption that messages would be of smaller lengths [4] . However, experiences reveal that messages are of varying length, majority of them being of short length. The message sizes range from a few bits used in acknowledgements to medium sized blocks, e.g. around 1 K bits, used to transmit long messages. A compromise in message size was therefore made for the public networks. It is now almost universally accepted that messages of 1000-2000 bits be used for the basic message unit or packet. In packet switched systems if a particular message eminating from a user is of longer length then the message is broken up into more than one packet for transmission. In message switching the messages are accepted in their entirety. Compared to message switching

where repeated storing and forwarding causes delay, the use of packets reduces the delay and also reduces the amount of storage needed. This is the main difference between packet switching and message switching. There are other numerous advantages of packet switched systems over circuit switched systems. To quote a few:

1) Circuit switching deals with information as binary streams at a prescribed speed while packet switching deals with information as short messages at any convenient speed. The speeds might vary from one part of the link to the other.

2) The standard which unites the source and destination of a"circuit" is its rate in bits per second while the standard which unites the source and destination of a "packet" is the maximum number of bits it can contain.

3) Multiplexing in circuit switching can be based on frequency or time division but this is irrelevent to the needs of a network user who has to deal with each circuit individually. Multiplexing in packet switching is by interleaving packets and it is easy for a programmed device to handle more than one packet stream at a time.

4) Circuit switching produces a small and constant transmission delay but packet delay is greater and variable.

5) Circuit switching requires a path to be found through the network before any data can pass but packet switching can begin to transport data at once. The only requirement is that the next node be available.

6) Packet switching is more economical because it interleaves packets from different streams according to their demand for channel capacity.

7) Packet switching requires the network to provide its own error control and therefore can improve its own bit error rate.

2.6 Design Considerations:

The concept of packet switching has been discussed earlier. To get the idea of a computer communication packet switched system we will outline the factors to be taken into consideration for the development of the communication network with packet switching. They are:

1) The design of Interface Message Processors (IMPs) to act as nodal store and forward switches.

2) The topological design to specify the capacity and location of each communication circuit within the network.

3) The design of higher level protocols for use of the network by time sharing, batch processing and other data processing systems.

4) System modelling and measurement of network performance and finding out the network parameters like delays, and throughput etc.

5) The reliability of the network

6) Responsiveness of the system

7) Routing

8) Flow control

9) Error control

10) Cost effectiveness of the system.

2.7 Hardware Problems:

As we have seen in the foregoing sections, the concept of packet switching is rather simple and its use fascinating. But its implementation has to cater for numerous hardware difficulties. Some of these are:

1. Co-ordination of the input-output is a must. One has to design suitable modems which can switch a slow rate input to a fast rate output and vice-versa.

2. Software and hardware are to be designed for routing packet(s) from one node to the other after analysing the network configuration and the channel behaviour.

3. Device to device signalling - to build a universal protocol for information exchange.
4. Error handling
5. Data format translation
6. Synchronisation
7. Memory and buffer sizes.

2.8 Definitions:

Following are the definitions of some of the terms associated with the packet switching system of computer communication.

Band:- The unit of signalling speed. It is the number of signal elements per second. Since a signal element can represent more than one bit, band rate is not the same as bit per second.

Buffer:- A store, usually associated with a peripheral device or communication line, which accomodates differences variation of data rate.

Capacity:- An abbreviation of 'channel capacity' which is the maximum rate at which data can be transmitted over the channel, measured in bits per second.

Channel:- A path along which signals carrying data are sent. Unless otherwise stated the term implies one-way communication

whereas the word circuit implies two way communication.

<u>Circuit switching</u>:- Switching as performed in the telephone network where communication is established, for the purpose, from one subscriber to the another, the circuit being held for the duration of the call.

<u>Cohesion</u>: The cohesion of a connected network is the least number of lines which must be removed to separate the network into more than one parts.

<u>Congestion</u>: The condition of a communication network beyond the limit of the traffic which it can readily handle, where there is a reduced quality of service and the network must restrict incoming traffic in order to remain effective. Congestion is taken care of by the flow control in a packet switched network.

<u>Connectivity</u>: The least number of nodes that must be removed to separate the network into more than one parts.

<u>Data</u>:- Digital information for processing, storage or transmission.

<u>Data rate</u>: The rate at which a channel carries data, measured in bits per second, known also as data signalling rate.

Demand Multiplexing: A form of time division multiplexing in which allocation of time to subchannels is made according to their need to carry data. A subchannel with no data to carry is not given any time slot. Another name for it is dynamic multiplexing.

Flow control: The control of data flow to prevent overspill of queues or buffers on loss or data because the intended receiver is unable to accept.

Host: A computer attached to a node in a communication network.

IMP: An interface message processor is the name given to a switching node.

Intellegent terminal: A terminal containing a microprocessor and therefore able to validate data implement protocols and so on,

Interface: A boundary between two parts of a system across which the interaction is fully defined. It includes a type of connector, signal levels, impedances and timing etc.

Message: Blocks of text or blocks of data which the user of a communication network wants to transport from one node to the other.

Modem: The device which (1) accepts a digital signal and transforms it into a form suitable for transmission over an analogue channel and (2) on receiving an analogue signal converts it back into a digital form.

Protocol: A strictly defined procedure for interaction across an interface or through a communication facility.

Queue: A store or buffer where packets come and wait for further processing.

Store-and-forward: The handling of messages or packets in a network by accepting them completely in a storage before forwarding them to the next node.

Topology of networks: The geographical layout of nodes and lines of a network.

CHAPTER 3

MESSAGE PATH DELAYS IN PACKET-SWITCHED COMMUNICATION NETWORKS

3.1 Introduction:

A communication path in a packet switched store-and-forward communication network is considered. A communication network is modelled by a weighted graph. The nodes on vertices represent the communication centres which correspond to switching centres in the network. The links or edges represent the communication channels. Channel capacity weights are assigned to the edges of the network. When a channel is busy the messages on packets directed into it join a queue which is governed by first come first serve algorithm or any other chosen algorithm. This is accomplished physically by storage of the messages or packets in buffers. The motion of queues of messages forming at the nodes is basic to the communication network under consideratio We may thus think of the communication nets as a network of queues. When the message reaches the front of the queue, th routing procedure is used to decide which channel the message will be sent over. Clearly, no other message or packet can be allowed to use the channel during that time. When its

transmission is completed this channel may accept a new message (packet) from the queue for transmission or it may serve for something else. The network is referred to as store -and-forward network because in passing through a node the packets are stored if necessary and then forwarded (transmitted) to the next node on its way to its destination. In transmission between two nodes a packet is considered to be received at the second node only after it is fully received. The consequences of this assumption is that a packet may not be transmitted further out of a node while it is being received at the node from a previous one.

Messages arrive at random at a source station (node) and follow a specific route in the network towards their destination. Message lengths are usually considered to be random variables. The origination time and lengths of the population of messages that will flow through the network cannot be predicted before hand with complete accuracy. We may however describe these random variables statistically by means of their probability density functions. Also, at a receiving node we assume that the arrival time and lengths of each such message are random. The node might be connected to more than one incoming line. From queueing

theory we further assume that inter-arrival times between messages are essentially distributed (i.e. Poisson) and so are the message lengths. Further more, we assume stationarity for these stochastic processes that feed the network.

In a packet-switching communication network the message is divided at the source station into fixed-length sub-messages called packets. Thus the random variable representing the population of messages gets transformed into another random variable representing the number of packets. These packets are then sent independently through the network towards the destination. At the destination, all the packets associated with a specific message are reassembled to be used or sent to a sink. With the help of a weighted graph a large varities of information transmission and processing networks can be described. In certain situations one can associate the communication channels with the vertices of the graphs and stations with its edges as in cases where time delays in the network are evaluated and the main time delay involved is associated with information processing in the station. The major considerations while analysing and synthesizing a communication network are those of congestion time delays and reliable transmission of information over the channel.

In a packet-switched communication network the designer is interested in

1. The average message delay
2. The total traffic handling capability of the network
3. The routing procedure
4. The flow control
5. The priority discipline
6. The storage capacity at each node
7. The channel capacity of each link
8. The topological structure of the network
9. The reliability
10. The total cost of the system.

In this chapter we derive exact expressions for the distribution of message delays over the communication paths. The latter chapters describe the optimisation of topological structures of the network and various reliability criteria. The assumptions that we make in arriving at various expressions of message delays and waiting times are:

1. Perfectly reliable nodes and channels
2. Noideal channels
3. Negligible cross-office delays compared to the channel transmission time.

4. Single destination for each message with no defection.

5. Infinite storage capacity at each node

6. Full reception of message before re-transmission

7. Poisson arrival statistics

8. Exponentially distributed message lengths

9. First come first served (FCFS) discipline.

Whereas assumptions 1 to 6 do not correspond exactly to the true situation in any real network they do describe idealisations which are reasonably close to reality. This would imply that impairments can be compensated suitably to make them close to the above assumptions; and such compensations are part of the network, Assumptions 7 or 8 are of a somewhat different nature. In particular they assign specific distributions to message arrival times and lengths. There is no available quantitative data from which to determine the actual form of these distributions. Moreover, these distributions avoid considerable mathematical complication.

For fixed message lengths exact impressions for waiting and delay time distributions along a communication path in a store-and-forward network have been derived in [5]. Approximate time delay results for store-and-forward network

assuming exponentially distributed message lengths are discussed in [6] and [7]. Many of the recent time delay reports have been associated with the Advanced Research Project Agency (ARPA) computer communication network.

Our delay analysis involves a single path in isolation, so that the messages in this path do not interfere wi any other message in the network. In the next section we present a mathematical model of a communication network and some definitions. Subsequent sections will cover message time delays over a single channel path and an 'N' channel path.

3.2 Mathematical Model and Definitions [8]

Figure 3.1 illustrates a path in a packet-switching communication network we consider this as our mathematical model for the calculations of various delays. The ith channel is represented by the edge (V_i, V_{i+1}) formed by connected vertices V_i and V_{i+1}. The capacity of ith channel is C_i, i=1,2,...N. The vertices represent stations or switchin centres or buffer systems equipped with memory storage units and queueing facilities.

Messages arrive at input station V_i at random times following a poisson stream with arrival intensity λ messages per second. The message length is assumed to be random.

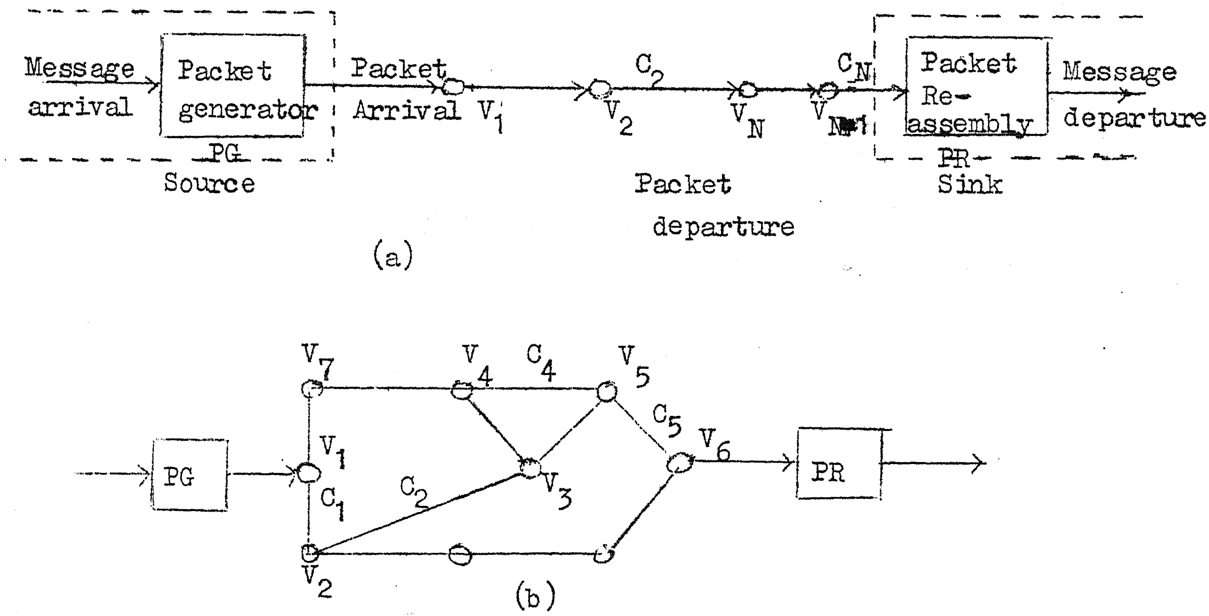


Fig. 3.1: (a) Communication path
(b) A packet switching communication network.

Let Y_n (bits) denote the length of the nth arriving message at V_i.

Assume $\{Y_n,\ n \geqslant 1\}$ to be independent identically distributed (iid) sequence of non-negative random variables governed by the distribution function $F_Y(y)$ where

$$F_Y(y) = P\{Y_n \leqslant y\} \tag{1}$$

Average message length μ^{-1} (bits per message) is

$$\mu^{-1} = E[Y_n] = \int_0^\infty y\ dF_Y(y) \tag{2}$$

Packet length of α (bits per packet) is fixed. M_n = number of packets representing nth message. $\{M_n, \ n \geqslant 1\}$ is a sequence of iid random variables following the probability measure g(m) where $g(m) \triangleq P\{M_n = m\}$

$$= F_Y(\alpha m) - F_Y[\alpha (m-1)]_{m \geqslant 1} \qquad (3)$$

$$\sum_{m=1}^{\infty} g(m) = 1$$

Average number of packets per message,

$$\bar{M} = \sum_{m=1}^{\infty} m\, g(m) \text{ packets per message} \qquad (4)$$

If the message length is exponentially distributed with mean μ^{-1}, we have,

$$F_Y(y) = \left[1 - e^{-\mu y}\right] u(y) \qquad (5)$$

where u(y) is a unit step function.

Using equation (3) we get,

$$g(m) = (1 - e^{-\mu\alpha m}) - (1 - e^{-\mu\alpha(m-1)}) \qquad (6)$$

$$= e^{-\mu\alpha(m-1)} - e^{-\mu\alpha m}$$

$$= e^{-\mu\alpha(m-1)} \left[1 - e^{-\mu\alpha}\right] \qquad (7)$$

Putting $B = e^{-\mu\alpha}$ and $A = (1-B)$ we get

$$g(m) = AB^{(m-1)} \tag{8}$$

The packets formed at V_i from each message are then transferred through the network to their destination in a store and forward manner. If a particular channel i is busy, the packets join the queue at V_i and are served on a first come first served discipline.

Each channel with its storage facilities can be considered as a queueing system. Let $P_t^{(i)}$ be the number of packets stored at V_i or being transmitted through the channel i at time t.

Therefore $\left\{ P_t^{(i)} , t \geqslant 0 \right\}$ is a queueing process associated with channel i.

We take $P_o^{(i)} = 0, \ i = 1,2,\ldots N$

The random instants of arrival of packets at V_i are denoted by,

$$\left\{ t_{n,j}^{(i)} , n = 1,2,\ldots, \ j = 1,2,\ldots M_n \right\}$$

The instants of arrival of nth message at V_i are given by

$$\left\{ T_n^{(i)} , n = 1,2, \ldots . \right\}$$

We see that,

$$T_n^{(i)} \triangleq t_{n,1}^{(i)}$$

The instants of packet departure from channel i is given by

$$\left\{ r_{n,j}^{(i)} \ , n = 1,2, \ldots, \ j = 1,2, \ldots M_n \right\}$$

The instants of message departure from channel i is given by

$$\left\{ R_n^{(i)} \ , n = 1,2, \ldots \right\}$$

We again see that,

$$R_n^{(i)} \triangleq r_{n,M_n}^{(i)}$$

If there is no transmission delay, we note that for a communication path,

$$t_{n,j}^{(i+1)} = r_{n,j}^{(i)}$$

and $T_n^{(i+1)} = R_n^{(i)} \quad , i = 1,2, \ldots N-1$

It means that the instant of departure of a message or a packet from channel i is the same instant of arrival of that particular message or packet at the node V_{i+1}.

The message/packet delay in a channel i is the total time the message/packet spends from the time it entered node V_i and the time it came out of the channel i or entered node V_{i+1}. The packet delay time is the sum of its waiting time and the transmission time in a channel.

Therefore,

Total delay time = waiting time + transmission time.

The waiting time at V_i of the jth packet associated with nth message is given by

$$\left\{ W_{n,j}^{(i)} \ , \ n = 1,2,\ldots, \ j = 1,2, \ \ldots M_n \right\}$$

The waiting time at V_i of nth message is $\bar{W}_n^{(i)} = W_{n,1}^{(i)}$.

The transmission time a_i of a packet of length over a channel i with capacity C_i is given by

$$a_i = \frac{\alpha}{C_i} \quad \text{sec. per packet} \tag{9}$$

The packet delay over the ith channel is given by

$$\gamma_{n,j}^{(i)} = W_{n,j}^{(i)} + a_i \tag{10}$$

The nth message delay time over the ith channel is given by

$$\bar{\gamma}_n^{(i)} = r_{M_n}^{(i)} - T_n^{(i)} \tag{11}$$

It is the time difference between the instant of departure of the last associated packet (M_n th) with nth message and the instant of arrival of the nth message at V_i.

Similarly the overall time delay of the nth message through the N channel communication path is given as the time difference between the instant of departure of the last associated packet from Nth channel and the instant of arrival of message at V_1.

Therefore,

$$\bar{\gamma}_n = r_{n,M_n}^{(N)} - T_n^{(1)} \qquad (12)$$

3.3 The Single Channel Path:

Before studying the delays over an N channel communication path we will derive some results for the single channel path and use them in the calculations for N channel path. Consider a single channel and consider packets arriving according to a poisson stream with intensity of messages per sec. Each of them requires a fixed transmission time or service of length a_1. Therefore for packet delays we need to consider a $M/D/1^*$ queueing system. However, as will be explained later, for message time delays we have to consider M/G/1 queueing system. We now need to obtain

the distribution of the message waiting time $\bar{W}_n^{(1)} = W_{n,1}^{(1)}$.
From the analysis given in [9 pp. 167], we can write,

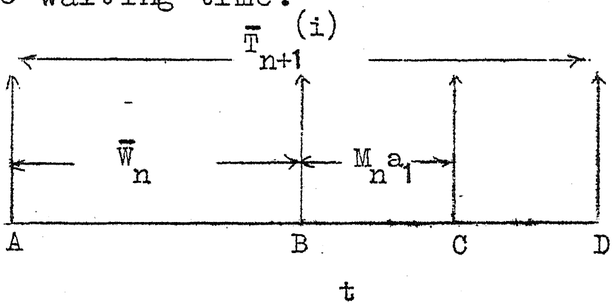
$$\bar{W}_{n+1}^{(1)} = \left[\bar{W}_n^{(1)} + M_n a_1 - \bar{T}_{n+1}^{(1)}\right]^+ \qquad (13)$$

where $[X]^+ \quad \max(0,X)$

and $\bar{T}_{n+1}^{(i)} = T_{n+1}^{(i)} - T_n^{(i)}$

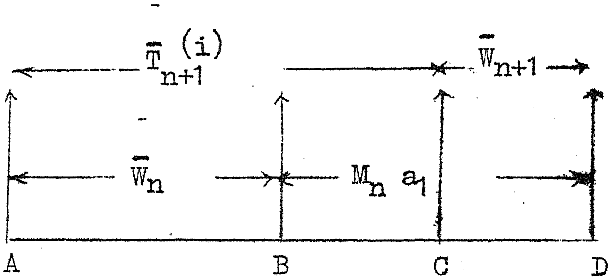
It denotes the interarrival time of $(n+1)$ message at channel i.

Figure 3.2 illustrates cases of zero waiting time and a finite waiting time.



(a) waiting time = 0

$\bar{T}_{n+1}^{(i)} \quad \bar{W}_n + M_n A_1$



(b) Waiting time = $\bar{W}_{n+1}$

$\bar{T}_{n+1}^{(i)} \quad \bar{W}_n + M_n A_1$

Figure 3.2 Waiting Time.

A - Instant of arrival of nth message

AB - Waiting time for nth message

BC - Service time for nth message

D - Instant of arrival of (n+1)st message

AD - Interarrival time of (n+1)st message

Therefore,

$$\bar{W}_{n+1} = \begin{cases} 0 & \text{if } R_n - T_{n+1} \leqslant 0 \\ T_n + M_n A_1 + \bar{W}_n - T_{n+1} & \text{if } R_n - T_{n+1} > 0 \end{cases} \tag{14}$$

---

*The notation A/B/S is commonly used to designate a queueing system with 's' servers with inter arrival distance 'A' or service times governed by 'B'. Thus an M/D/1 queueing system is a 1 server system with Poisson arrival or deterministic service times Kendall,D.G. 1951 introduced a short hand notation for describing which type of queueing situation is meant. The notation consists of 3 symbols, the first referring to the type of interarrival time distribution. The second to the service time distribution and the third to the number of servers. In this notation which is now in common use, the symbol

M stands for the negative exponential distribution;

Poisson stream

E stands for enlarg distribution

H stands for hyperexponential distribution

D stands for deterministic value

G stands for general distribution

Relation 13indicates that for deriving the message waiting times, we need consider a M/G/I queueing system with unit Poisson arrivals of intensity λ and service times equal to $M_n a_1$. The waiting time sequence $\{\bar{W}_n, \ n \geqslant 1\}$ is governed by the same statistics as the corresponding sequence for a M/G/1 queueing system with Poisson arrivals of iid service times $\{X_n, n \geqslant 1\}$. The limiting waiting time distribution as given in [9 pp. 253-256] is

$$W^{(1)}(t) \triangleq \lim_{n \to \infty} (\bar{W}_n^{(1)} \leqslant t)$$

$$= (1-\rho) \sum_{n=0}^{\infty} \rho_1^{\,n} \left[(\bar{a}_1)^{-1} \int_0^t \{1-B(\tau)\} d\tau \right]^{n*}, \ t > 0 \tag{15}$$

where $B(t) = \{X_n \leqslant t\} = \sum_{m=1}^{t/a} g(m)$; $[x]$ is the largest integer not larger than x.

$F(t)^{n*}$ denotes the nth convolution of $F(t)$

$\rho_1 = \lambda E(M_n) a_1$

$\bar{a}_1 = E(M_n) a_1$

The limiting message mean waiting time is given by

$$W^{(1)} \quad \operatorname{Lim} E\left\{ \bar{W}_n^{(1)} \right\} = \frac{1}{2}\left(1+CM^2\right) \frac{\rho_1 \bar{a}_1}{1-\rho_1} \qquad (16)$$

where

$$CM^2 = \frac{\operatorname{Var}(M_n)}{\left[E(M_n)\right]^2} \qquad (17)$$

ρ is the utilisation factor i.e. the fraction of time that the channel is busy with the customers. For stable operation

$\rho < 1$

$$\rho = \frac{\lambda}{\mu C}$$

D Message delay time as defined by (11) can be written as

$$\bar{\gamma}_n = \bar{\gamma}_n^{(1)} = \bar{W}_n^{(1)} + M_n a_1$$

The steady state distribution of the message delay (D_m) is given by

$$D_m = \lim_{n\to\infty} E\left[\bar{\gamma}_n\right] = \lim_{n\to\infty} E\left[\bar{W}_n^{(1)}\right] + E\left[M_n a_1\right]$$

$$D_m = \frac{1}{2} \frac{\rho\, \bar{a}_1}{1-\ _1} \left(1+C_n^{\,2}\right) + \bar{a}_1 \qquad (18)$$

.4 N-Channel Communication Path:

Consider an N-channel path where $N \geqslant 2$. Due to the service structure of the path, we have $t_n^{(i)} = r_n^{(i-1)} \quad i \geqslant 2$,

so that the interarrival time at channel i, i 2 is given by

$$T_{n+1} = T_{n+1}^{(i)} - T_n^{(i)} = R_{n+1}^{(i-1)} - R_n^{(i-1)} \tag{19}$$

$$= \begin{cases} M_n a_{i-1} & \text{if } \bar{W}_{n+1}^{(i-1)} > 0 \\ M_n a_{i-1} + I_{n+1}^{(i-1)} & \text{if } \bar{W}_{n+1}^{(i-1)} = 0 \end{cases} \tag{20}$$

where $I_{n+1}^{(i-1)}$ denotes the duration of ideal period prior to $t_{n+1,l}^{(i-1)}$ at channel (i-1)

The waiting time at channel (i) follows the relationship (13)

$$\bar{W}_{n+1}^{(i)} = \left[\bar{W}_n^{(i)} + M_n\, a_i - \bar{T}_{n+1}^{(i)}\right]^+ \tag{21}$$

Hence by eqn. (20) and (21) we conclude that

$$\bar{W}_n^{(i)} = 0, \text{ if } a_i \leqslant a_{i-1}, \quad i \geqslant 2 \tag{22}$$

As in [Theorem 697 pp. 218-219 of '5'] The overal delay time over an N channel path with capacities $(C_1, C_2 \ldots C_N)$ is the same as that over an N channel path with capacities $(C_{i1}, C_{i2}, \ldots C_{iN})$ where the latter sequence is an arbitrary ordering of $(C_1, C_2, \ldots C_N)$. The overall waiting time is invariant to the order of channels and depends only on the minimal capacity min $(C_1, C_2, \ldots C_N)$.

Therefore we can say that,

$$\gamma_{n,M_n}^{(N)} \quad \gamma_{n,1}^{(N)} = (M_n - 1)\, a_{max} \tag{23}$$

where $a_{max} = \alpha / C_{min}$ or max. $(a_1, a_2, \ldots a_N)$.

The overall all time delay of the nth message through the N-channel communication path as defined by (12) is

$$\bar{\gamma}_n \triangleq \gamma_{n,M_n}^{(N)} - t_{n,L}^{(1)}$$

$$= \gamma_{n,L}^{(1)} - t_{n,1}^{(1)} + \gamma_{n,M_n}^{(N)} - \gamma_{n,1}^{(1)} \quad \text{(By adding and substracting } \gamma_{n,1}^{(N)}\text{)}$$

$$= \sum_{i=1}^{N} \bar{W}_n^{(i)} + \sum_{i=1}^{N} a_i + (M_n - 1)\, a_{max} \qquad (24)$$

As in (16) the limiting average waiting time over the N-channel path is given by

$$W \triangleq \lim_{n \to \infty} E\left(\sum_{i=1}^{N} \bar{W}_n^{(i)} \right)$$

$$W = \frac{1}{2} \frac{\rho_{max}\, \bar{a}_{max}}{1 - \rho_{max}} (1 + C_M^2) \qquad (25)$$

where C_M^2 is given by (17), $\bar{a}_{max} = E[M_n]\, a_{max}$, $\rho_{max} = \lambda \bar{a}_{max}$.

The overall average time delay over the path is

$$D_m \triangleq \lim E(\bar{\gamma}_n)$$

From (24) and (25)

$$D_m = \frac{1}{2} \frac{\rho_{max}\, \bar{a}_{max}}{1 - \rho_{max}} (1 + C_M^2) + \sum_{i=1}^{N} a_i + E\left[(M_n - 1)\, a_{max} \right]$$

$$D_m = \frac{1}{2} \frac{\rho_{max}\,\bar{a}_{max}}{1-\rho_{max}} (1+C_M^2) + \sum_{i=1}^{N} a_i + \bar{a}_{max} - a_{max} \qquad (26)$$

Fig.3.3 shows the variation of the message overall delay time as a function of traffic intensity ρ. For a relative study we have taken exponentially distributed message lengths with mean μ^{-1} and equal channel capacities $C_i=1$, $i=1,2,\ldots N$. To make the plot independent of N we take the delay $\gamma = D_m-(N-1)a_i$. D_m is given by (26). The figure indicates that for low traffic intensities one can choose longer packet length without causing too high a resulting delay. For higher intensities the choice of packet length is governed by allowable path delay.

Fig.3.4 is plotted for the practical values of μ, A and α. We see that the overall delay is within 2 sec. which is allowable message delay in most of the existing networks.

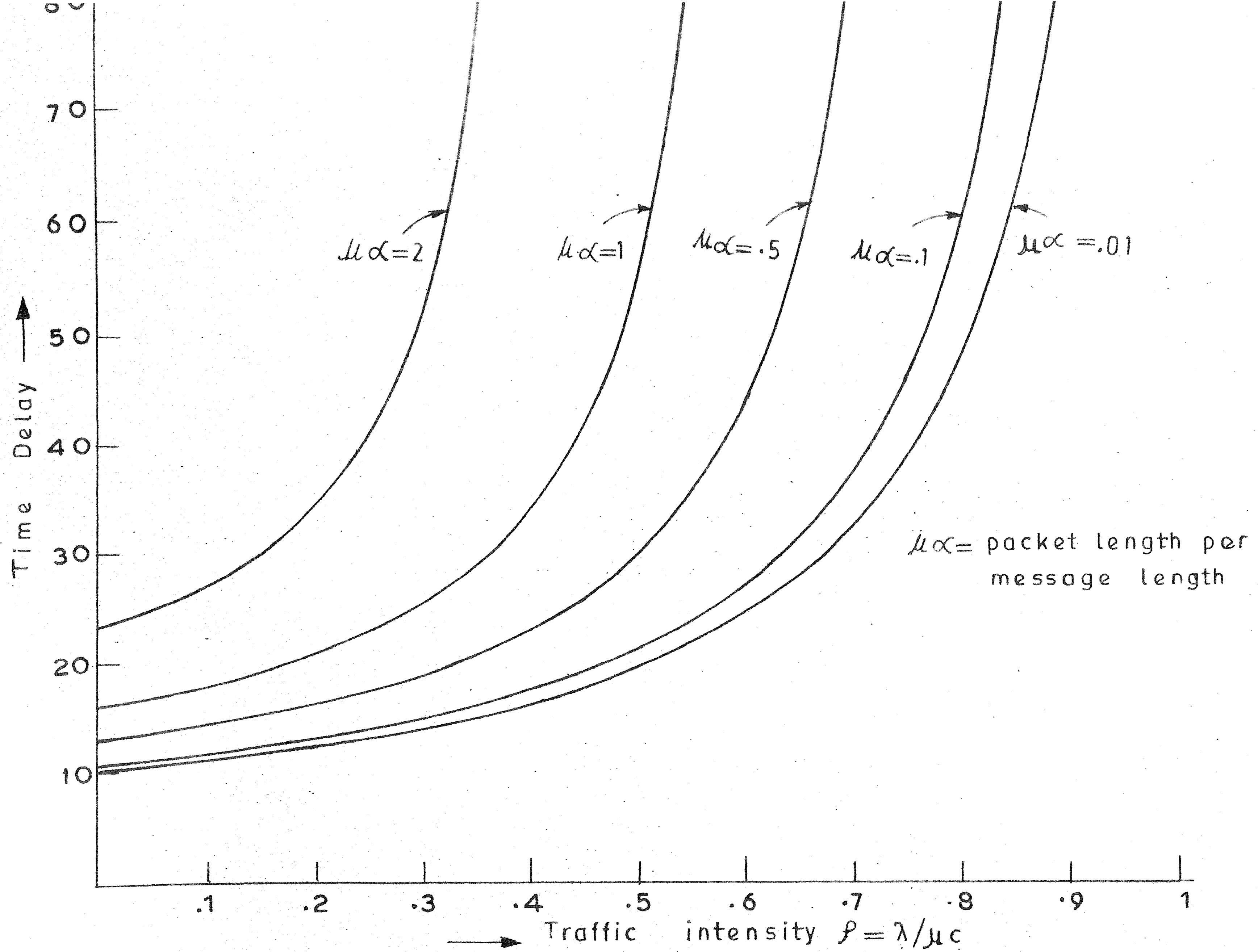


FIG.3.3 Delay versus traffic intensity for different $\mu\alpha$

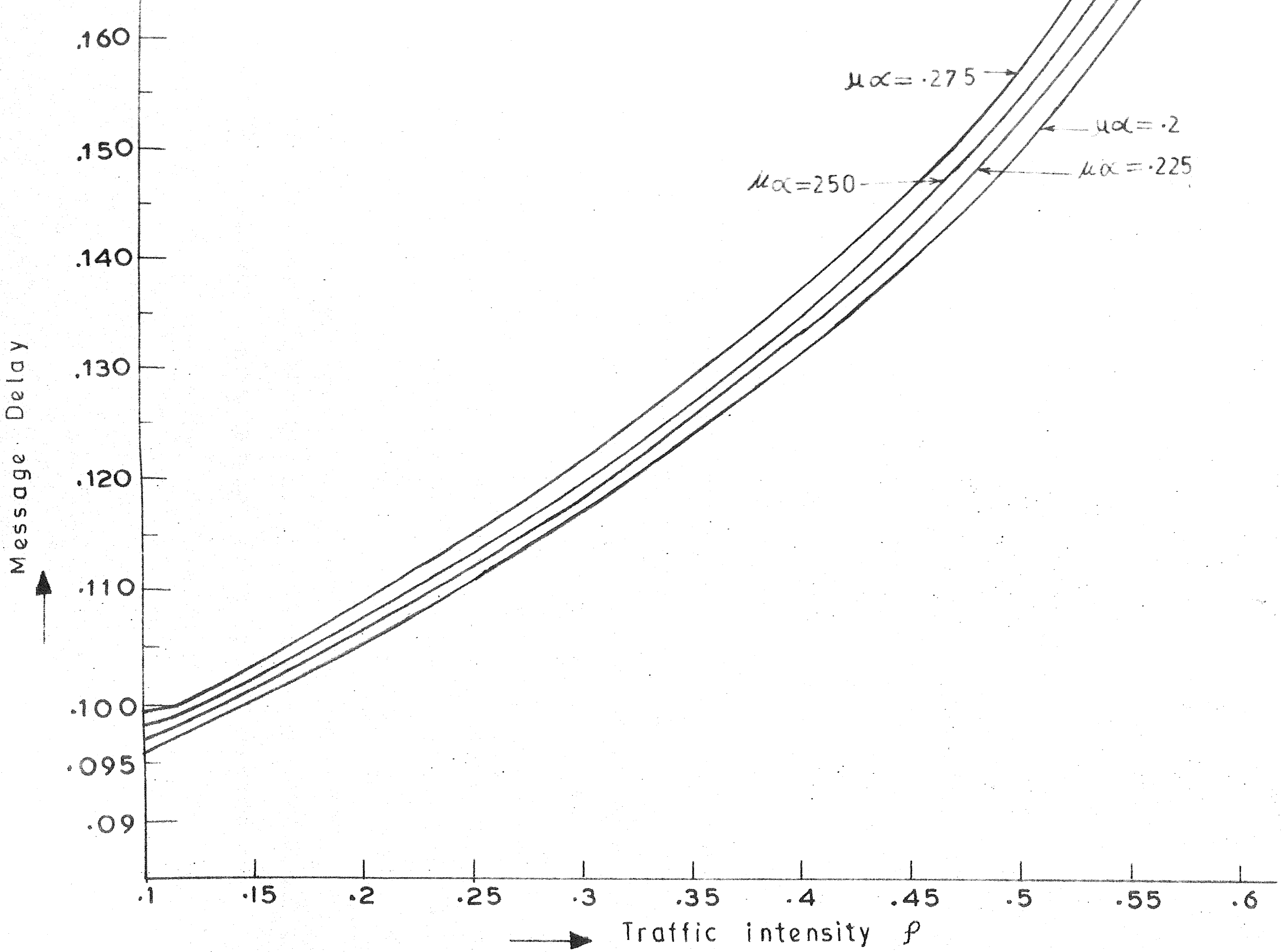


FIG.3.4 Delay versus traffic intensity for different packet lengths

CHAPTER 4

RELIABILITY OF A NETWORK

4.1 Introduction:

In communication engineers' parlance a communication system should perform well to the satisfaction of users. In digital terms, this performance measure is generally the probability of error (bit error rate) which gives the tolerance bound on the performance. Analogously, in computer communication, we are interested in establishing apparent or direct link between two nodes through a network which should remain operational to the satisfaction of users. The measure is generally called the reliability measure which is a probabilistic measure that tells us whether a system will perform adequately under the system constraints. Obviously, the reliability of a computer communication system would depend on the hardware reliability (most commonly expressed as communication problem) as well as the software reliability and network reliability. As the communication networks become more complex, the complexity of reliability models also increase. Network reliability would depend on the availability of the communication paths between all pairs of centres in the network, i.e. on topological considerations.

There have been many definitions of the term reliability and the reliability measures have been expressed in terms of different parameters. One definition of reliability is the ability of the communication system to function even after partial failures. Alternatively a network is considered operational in the presence of failures of nodes or links if every node could communicate with a certain number of other nodes. In general we can say that the reliability of a communication system is the ability of the system to communicate between two points without interruption over a specified period of time. The various methods for finding the reliability of a communication network ultimately boil down to finding a measure which describes the number of links or modes that will have to be removed to disrupt the communication between two points. In this chapter we shall use graph theoratic approach for the consideration of reliability of a communication network.

4.2 Reliability Criteria:

The reliability criteria can be categorised as either to be deterministic or probabilistic. The deterministic criteria were originally formulated as vulnerability measures for the communication networks. This criterion assumes importance when the aim is the deliberate destruction

of a communication network. It is assumed that the destructing force has the complete knowledge of the topology of the network. Therefore, the deterministic criteria indicates how difficult it is for the destructing force to disrupt the network. It is a measure of the number of nodes or links that would have to be destroyed to completely disrupt the network.

Probabilistic criterion on the other hand are considered when the elements of the network are destroyed at random. The destructing force does not have the knowledge of the topology of the communication network. Such criterion were originally introduced in a military environment where the enemy would be interested in destroying the communication installations at random. Natural calamities such as earthquakes, cyclones and other such accident also contribute to the destruction of nodes or links of a communication network at random.

In designing maximally reliable communication networks, subject to fixed cost constraints, the objective is to maximise the probability that the network is operational based on a specified probability of failure for every node or link in the network. The exact calculations of the reliability of the communication path between any pair of nodes

in a distributed computer communication network have not been feasible for large network because of complexity involved and consequently many reliability criterian have been suggested for a good approximation to the network reliability.

4.3 Reliability Measures:

Based on the various definitions of reliability and the approximations to the complexity of a communication network, various reliability measures have been suggested. One can adopt any one of them depending upon the complexity of the network. The various reliability measures suggested are based on;

1) Cohesion of network
2) Connectivity of network
3) Generalised cohesion
4) Network diameter
5) Node pain failure probabilities
6) Cut sets.

Each one of the above methods has its own advantages and disadvantages and different approximations to the system reliability. We have analysed the reliability based on the minimal cut sets. The system reliability for two network models has been calculated for different probabilities of

success of individual links. Graphs have been plotted for system reliability versus the reliability of individual links. The reliability of each link has been considered to be the same for the ease in plotting of graphs. In a practical network one link may be more vulnerable than the other.

4.4 Definitions:

Computer network is modelled by a graph G = (N,E) where N is a non-empty set of 'n' nodes and E is a set of 'c' edges.

Degree of a node or valence di is the number of edges adjacent to it. We have

$$\sum_{i=1}^{n} di = 2c$$

Length of the path is the number of edges constituting the path.

Disjoint paths - Any path between nodes n_1 and n_2 are disjoint if they do not have any nodes in common except nodes n_1 and n_2.

Adjacent nodes are those nodes which have a common link.

Adjacent edges are those edges which have one node in common.

Node cut-set ; set of nodes the detection of which will disconnect the network.

Line cutset or cutset:- A set of edges the deletion of which will disconnect the network.

Minimal cutset:- A minimum set of edges the deletion of which will disconnect the network.

Connectivity:- The minimum size of node cutset.

Cohesion:- The minimum size of line cutset.

Diameter:- The maximum length of any shortest path in a network.

Nodes are the switching centres with store and forward facilities or may be a computer system in a computer communication network.

Edge: It corresponds to a link or a channel in a network.

4.5 System Reliability based on Minimal Cutsets:

As system technology becomes more sophisticated and reliability engineering is more widely practiced, the complexity of reliability models increases. It becomes difficult to analyse in detail a complex reliability structure. One of the difficulties associated with obtaining reliability

estimates for complex system is that of deriving a prediction equation which expresses all possible events of interest. One way to simplify this difficulty is to obtain a sequence of prediction equations which provide increasingly close bounds on the system reliability. In this chapter we are presenting a general topological approach based on graph theory and develop a bound and approximations which allow a considerable computer savings [12,15]. Conceptually the task of determining the system reliability is a simple one. The method most often suggested to determine the system reliability is to draw a reliability network graph, enumerate from the network all mutually exclusive working states of the system, calculate the probability of occurrence of each state and sum these probabilities. For a complex system this is not a practical method, because there is a very large number of working states i.e. the number of communication paths between two nodes of a network. For instance a ten component system will have $(2^{10}-1)$ states. The reliability of this can be calculated by a computer but a moderate increase in the system complexity or the number of components soon overcomes even the capacity of the computer.

This computational problem is considerably reduced by approximation techniques [11, 12], which by considering a

much smaller set of states called the set of minimal cuts obtains a good lower bound to the system reliability.

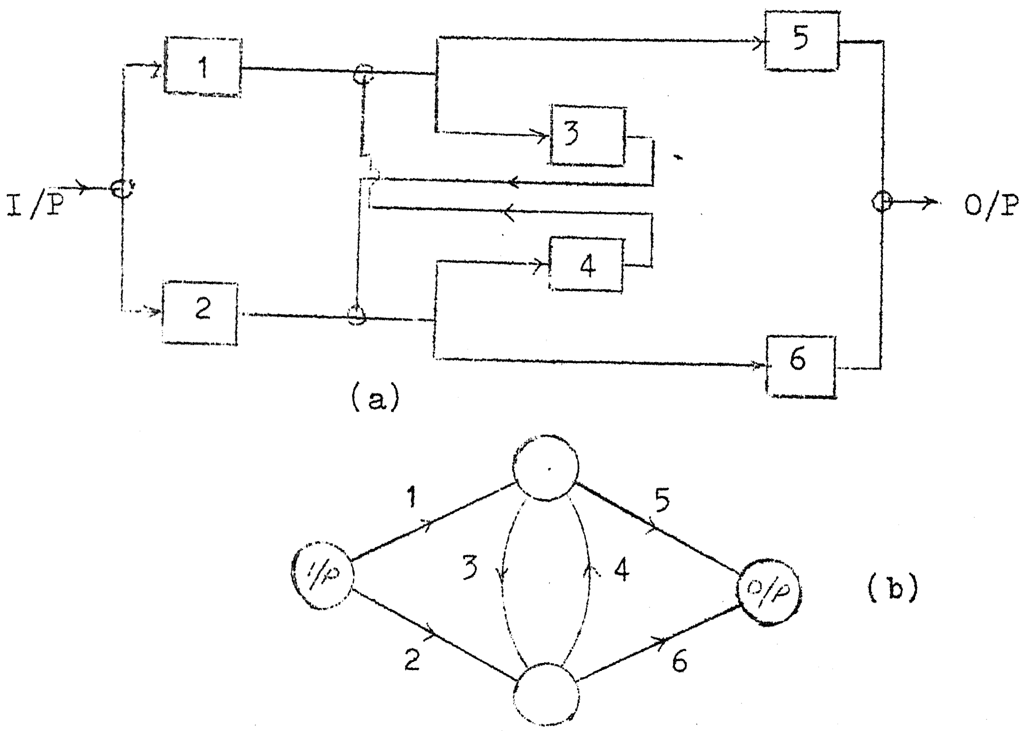


Fig. 4.1: (a) Simple functional logical diagram
(b) Reliability graph corresponding functional logical diagram.

Illustrating with a simple example, the system is represented by a reliability diagram or a graph as shown in Fig. 4.1. The rectangles represent components or the links of the system. Component i(i=1,2,...) has reliability or probability of success p_i which is known. It is one minus the probability of outage of each link. The reliability graph for the system

is disected into its constituent tie sets and minimal cutsets. As defined earlier, a minimal cut-set is a minimal set of elements which literally cuts all success paths in the graph and a tie set (success path) is a directed path between the input and the output. This is illustrated in Fig. 4.1(b). A minimal cutset will disrupt the communication between two nodes. For the example shown in Fig. 4.1 the tie sets are (1,5), (2,6), (1,3,6), (2,4,5) and the minimal cutsets are (1,2), (2,5), (1,4,6), (2,3,5).

Let T_i, i=1,2,...I denote the tie sets in a graph

Let C_j, j=1,2,...J denote the minimal cutsets

The reliability R of the system is the probability that atleast one tie set is operational or alternately it is the probability that all cut-sets are operational.

A tie set is operational if all its components provide a communication path between the input and the output.

A cutset is operational if atleast one of its component provides a communication path between the input and the output.

Therefore, the system reliability can be expressed as

$$R \triangleq P\left[T_1 + T_2 + \dots T_I\right] \qquad (1)$$

$$= P\left[\text{atleast one tie set is operational}\right]$$

For the example of Fig. 4.1 the reliability as expressed by (1) is

$$R = P\left[1.5 + 2.6 + 1.3.6 + 2.4.5\right]$$

Alternately the system reliability can be expressed as the probability that all minimal cuts are operative. We then obtain,

$$R = P\left[C_1, C_2, C_3 \ldots C_J\right] \qquad (2)$$

We can also say that

$$R = 1 - P\left[\bar{T}_1 . \bar{T}_2 \ldots T_I\right] \qquad (3)$$

It is the probability that all tie sets are bad.
Similarly

$$R = 1 - P\left[\bar{C}_1 + \bar{C}_2 + \ldots \bar{C}_J\right] \qquad (4)$$

$\bar{T}_i$ and $\bar{C}_j$ are the compliments of T_1 and C_J respectively.

Bounds for the system reliability can be obtained by using the basic probabilistic inequalities given below;

$$R = P\left[T_1 + T_2 + \ldots T_I\right] \leqslant \sum P\left[T_i\right] \qquad (5)$$

$$R = P\left[T_1 + T_2 + \ldots T_I\right] \geqslant \sum P\left[T_i\right] - \sum P\left[T_i . T_k\right].$$

$$1 \leqslant i, \quad K \leqslant I \qquad (6)$$

Indexing $1 \leqslant i$, $K \leqslant I$ is to ensure that a given set is not counted more than once.

Therefore the upper and lower bounds for the reliability are

$$R_U = \sum P[T_i] \tag{7}$$

$$R_L = \sum P[T_i] - \sum_{i<k} P[T_i \cdot T_k]$$

$$1 \leq i, \quad K \leq I \tag{8}$$

Since $\sum P[T_i]$ can be greater than 1, the modified upper bound is

$$R_U = \sum P[T_i] - \sum P[T_i \cdot T_k] + \sum_{i \leq k \leq l} P[T_i \cdot T_k \cdot T_l] \tag{9}$$

$$1 \leq i, \quad k, l \leq I$$

The last two summations are overall possible combinations of L,K,l.

Similarly, the bounds can be obtained using cutsets from (4)

$$R_{L1} = 1 - \sum P[\bar{C}_j] \tag{10}$$

$$R_{V1} = 1 - \sum P[\bar{C}_j] + \sum_{j<m} P[\bar{C}_j \cdot \bar{C}_m] \tag{11}$$

These bounds converge to the reliability which would be calculated if all the terms in the model are evaluated.

From the analysis given in [12 pp. 294-295] it is observed that the approximations involving tie sets (bounds) are useful in the low reliability region and those

involving cutsets are useful in the high reliability region which is generally the case in practical systems. The computations done in this chapterare are based on cutsets. The extent of computational saving done using these bounds for the reliability can be seen from the following. For a system of 'm' minimal cutsets and 'l' minimal tie sets:

R_U acquired l terms

R_L requires $\frac{l(l+1)}{2}$ terms

The complete tie set expansion required (2^l-1) terms i.e. equation (1).

Similarly R_{UL} requires 'm' terms

R_{L1} requires $\frac{m(m+1)}{2}$ terms.

Equation (4) requires 2^m-1 terms.

The reliability graphs for the networks of Fig. 4.1 and 4.2 are shown in Fig. 4.3. Network shown in Fig. 4.2 is more complex than that of Fig. 4.1 and needs computer calculations for the system reliability.

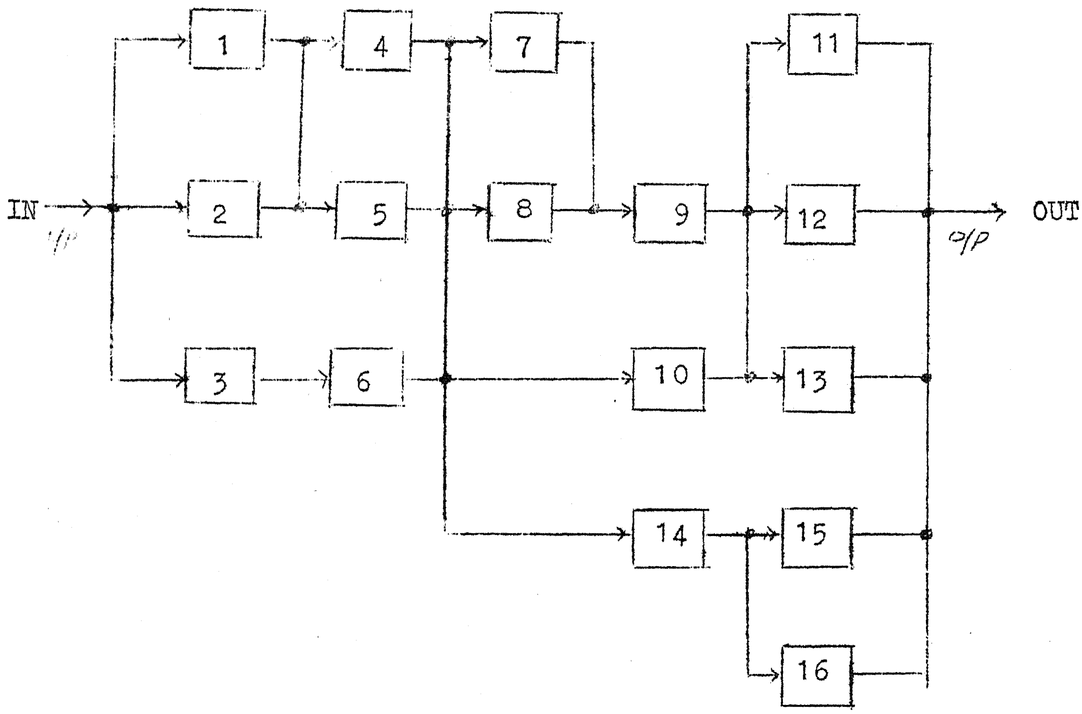


Fig. 4.2 Network Model for Reliability

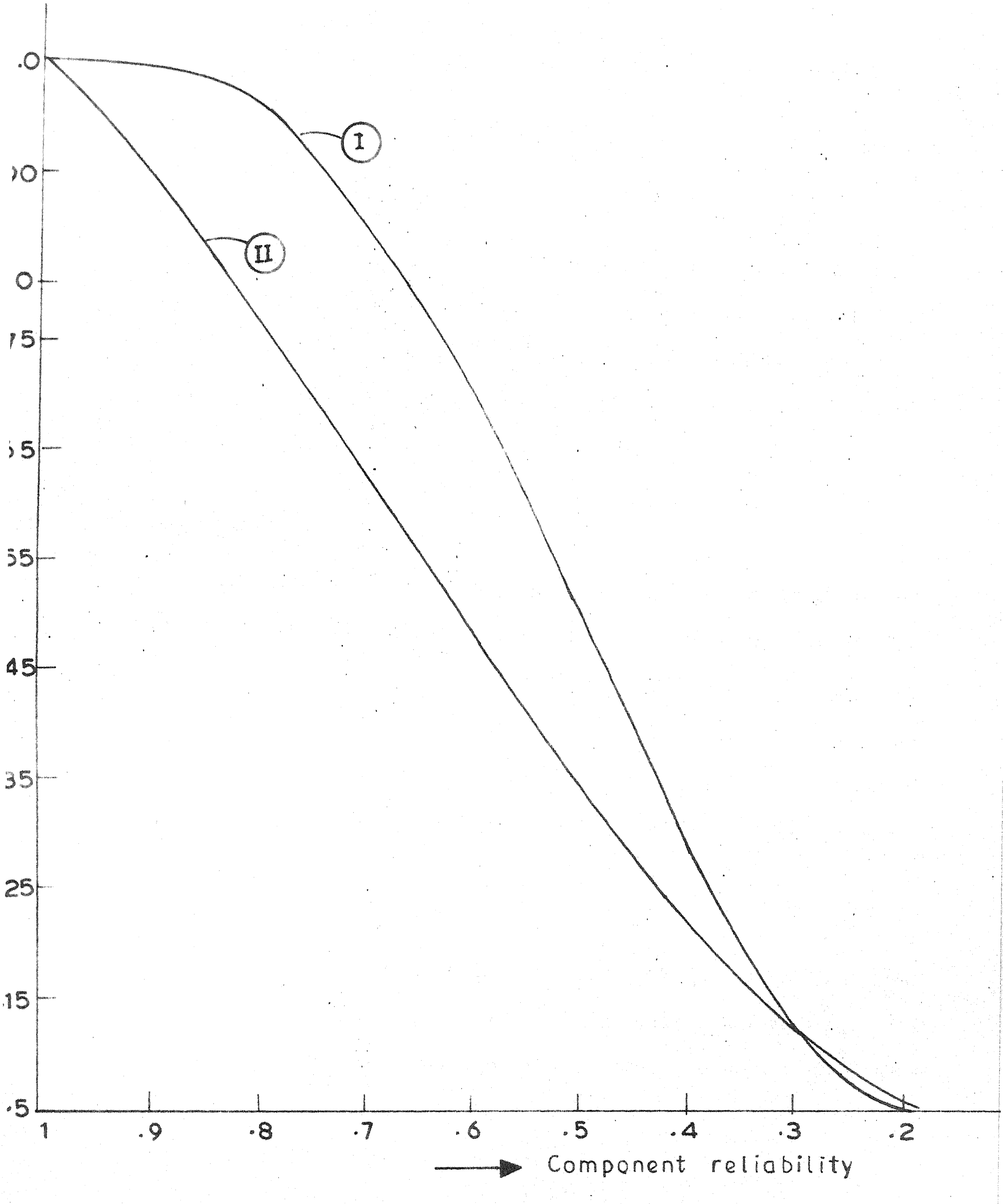


FIG.4.3 System Reliability versus Prob. of success of

CHAPTER 5

AVERAGE PATH LENGTH IN A NETWORK

5.1 Introduction:

The major factors to be taken into consideration while designing a computer communication network are the message delays, reliability, average path length and the cost. While the message delays and reliability have been discussed in Chapters3 and four respectively the cost factor varies from place to place and time to time. In this chapter the lower bounds on transmission delay have been established by measuring the average path length. The total delay associated with a packet in a packet switched network is because of the waiting time and the transmission time. The waiting time corresponds to the queueing delay and the transmission time depends on the length of the communication path between two nodes. Therefore minimising the number of intermediate nodes and the communication links through which packets have to pass to reach their destination, will reduce delay.

If every node in a N-node network is connected with every other node, it is a maximally connected graph. The shortest and average path length is one. However the number

of edges in network is $\frac{N(N-1)}{2}$. This network is the most reliable but the cost of transmission links increase as the square of the number of nodes. In a star configuration (discussed in Chapter 2) the shortest path between all nodes is two and the cost of communication links increases linearly with the number of nodes. But the reliability and capacity of the central node poses problems. These are the extreme cases in a network design. Therefore the prime interest is to keep the average path length as small as possible and the valence between two and (N-1).

There are several possible strategies for reducing the transmission delay in a network but in this chapter the delay because of the topological structure of the network has been examined.

5.2 Network Graph Description:

A packet switching network is modelled with a linear regular graph and we examine the network topology which produces minimum average path length. A relationship between valence, average path length and number of nodes is developed. This relationship will give us the average path length for a given number of transmission links in the network. Thus we will have a measure of the cost which depends on the number of

under a suitable delay constraint and reliability. The expression developed is in the form of lower bound on average path length.

Let us consider a tree with $N = 1 + V\left[\sum_{j=0}^{m-1} (V_1 - L)^j\right]$ nodes [16] $m \geqslant 1$ in which each node has valence V, (except the end nodes which have valence 1); 'm' is the level of node. For example a tree with V=3 and m=2, shown in Fig. 5.1, has 10 nodes. One node is chosen as the root

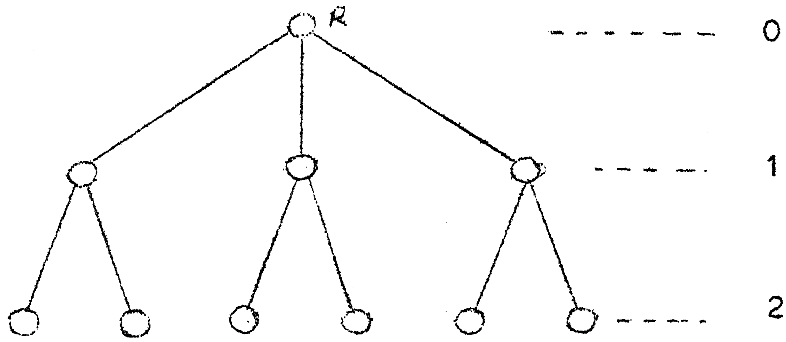


Fig. 5.1 A Typical Tree

node and is labelled as R. The root node R is considered to be at distance zero from itself in the tree. 'V' nodes (in this case 3) adjacent to 'R' are at distance 1 from R. The V(V-1) nodes adjacent to those at distance 1 from 'R' are at distance 2 from R and so on. For ease of expression, nodes at distance m from R will be referred to as 'nodes

at level m'. A tree so described is a complete tree since new nodes can only be added by starting a new level. In order to find the average path length from R to all nodes in the tree it is necessary to sum all paths and then divide by (N-1). It is obvious that there are V paths of length 1, V(V-1) paths of length 2, V(V-1)(V-1) paths of length 3 and so on.

5.3 Average Path Length:

The average path length L of a complete tree with 'm' levels is

$$L = \frac{1}{(N-1)} \left[V \sum_{j=0}^{m-1} (V-1)^{j} (J+1) \right] \qquad (1)$$

Now if we remove monovalent nodes (V=1) at the highest numbered level and their associated edges in the tree and if the number of vertices left in the tree is 'K' then the number of vertices removed is:

$$1 + V \sum_{j=0}^{m-1} (V-1)^{j} - k \qquad (2)$$

The average path length P(K,V) in this pruned tree is

$$L(K,V) = \frac{1}{K-1} \left[V \sum_{j=0}^{m-1} (V-1)^{j} (J+1) - \left(1 + V \sum_{j=0}^{m-1} (V-1)^{j} - K\right) m \right]$$

(3)

$$= \frac{1}{K-1}\left[V \sum_{j=o}^{m-1} (V-1)^{j}(J+1-m)+(K-1)m \right] \qquad (4)$$

From Eqn. (2) we know that

$$1 + V \sum_{j=o}^{m-1} (V-1)^{j} - K \geqslant 0 \qquad (5)$$

It can be shown that

$$\sum_{j=o}^{m-1} (V-1)^{j} = \frac{(V-1)^{m} - 1}{V-2} \qquad (6)$$

$$S = 1+(V-1)+(V-1)^{2}+(V-1)^{3}+ \dots (V-1)^{m-1}$$

$$S(V-1) = (V-1) \ (V-1)^{2}+ (V-1)^{3}+ \dots (V-1)^{m-1} +(V-1)^{m}$$

$$-S+S(V-1) = -1 + (V-1)^{m}$$

$$S(V-1-1) = (V-1)^{m} + (-1)$$

$$S = \frac{(V-1)^{m}-1}{V-2}$$

Inequality (5) can be written

$$1 + \frac{V}{V-2}\left[(V-1)^{m}-1 \right] - K \geqslant 0 \qquad (7)$$

m is the smallest integer satisfying the above inequality. Therefore,

$$(V-1)^{m}-1 \geqslant \frac{K(V-2)}{V}$$

$$(V-1)^{m} \geqslant \frac{(K-1)(V-2)}{V} + 1$$

$$= \frac{K(V-2)-V+2+V}{V}$$

If $\{y\}$ denotes the least integer $\geqslant y$ then

$$m = \left\{ \log_{(V-1)} K \frac{(V-2)+2}{V} \right\} \quad , \text{ for } V > 2 \qquad (8)$$

We can show that

$$V \sum_{j=o}^{m-1} (V-1)^{j} (J+1) = \frac{V}{(V-2)^2} \left[m(V-1)^{m+1} - (m+1)(V-1)^{m} + 1 \right] \qquad (9)$$

If we substitute for m, it is possible to obtain a more explicit relation between the average path length L(K,V), the number of nodes K and valence V.

Using (4), (6) and (9) we can write

$$L(K,V) = \frac{1}{K-1} \left[V \left\{ \frac{m(V-1)^{m-1} - (m+1)(V-1)^{m} + 1}{(V-2)^2} \right\} - mV \left\{ \frac{(V-1)^{m} - 1}{V-2} \right\} + (K-1)m \right] \qquad (10)$$

$$= V \left[\frac{1-(V-1)^{m} + m(V-2)}{(V-2)^2 (K-1)} \right] + m \qquad (11)$$

$$= \frac{V}{K-1} \left[\frac{1-(V-1)^{m} + m(V-2)}{(V-2)^2} \right] + m \qquad (12)$$

In eqn. (12) we substitute the value of m from eqn. (8).

$$L(K,V) = \frac{V}{(K-1)(V-2)^2}\left[1-(V-1)^{\left\{\log_{V-1}\frac{K(V-2)+2}{V}\right\}} + (V-2)\left\{\log_{V-1}\frac{K(V-2)+2}{V}\right\}\right] + \log_{V-1}\frac{K(V-}{V} \tag{13}$$

The average path length form is primarily logarithmic in K except for the case V=2 in which it is linear. Curves of L(K,V) for $2 \leqslant V \leqslant 6$ and K are shown in Fig. 5.2. Table 5.1 shows values of average path length for various V.

We note the drastic reduction in L(K,V) between the cases V=2 and V=3. But an increase to V=6 for K=900 only reduces the L(K,V) by a factor of 2 while it increases the number of edges in the graph from 1350 for V=3 to 2700 for V=6 . Expression (13) relates to the minimum average path length for all graphs with valence V.

Table 5.1

Some Typical Values of L(K,V)

		V				
		2	3	4	5	6
K	10	2.78	1.67	1.66	1.44	1.33
	20	5.26	2.37	1.95	1.74	1.68
	30	7.76	2.86	2.31	1.97	1.79
	40	10.26	3.15	2.49	2.23	1.85
	50	12.76	3.41	2.59	2.39	2.14
	100	25.25	3.21	3.27	2.70	2.58
	200	50.25	5.14	3.83	3.32	2.79
	300	75.25	5.80	4.22	3.55	3.24
	500	125.25	6.52	4.54	3.88	3.54
	700	175.25	6.94	4.98	4.20	3.67
	900	225.25	7.32	5.20	4.38	3.75
	1000	250.25	7.49	5.28	4.44	3.77

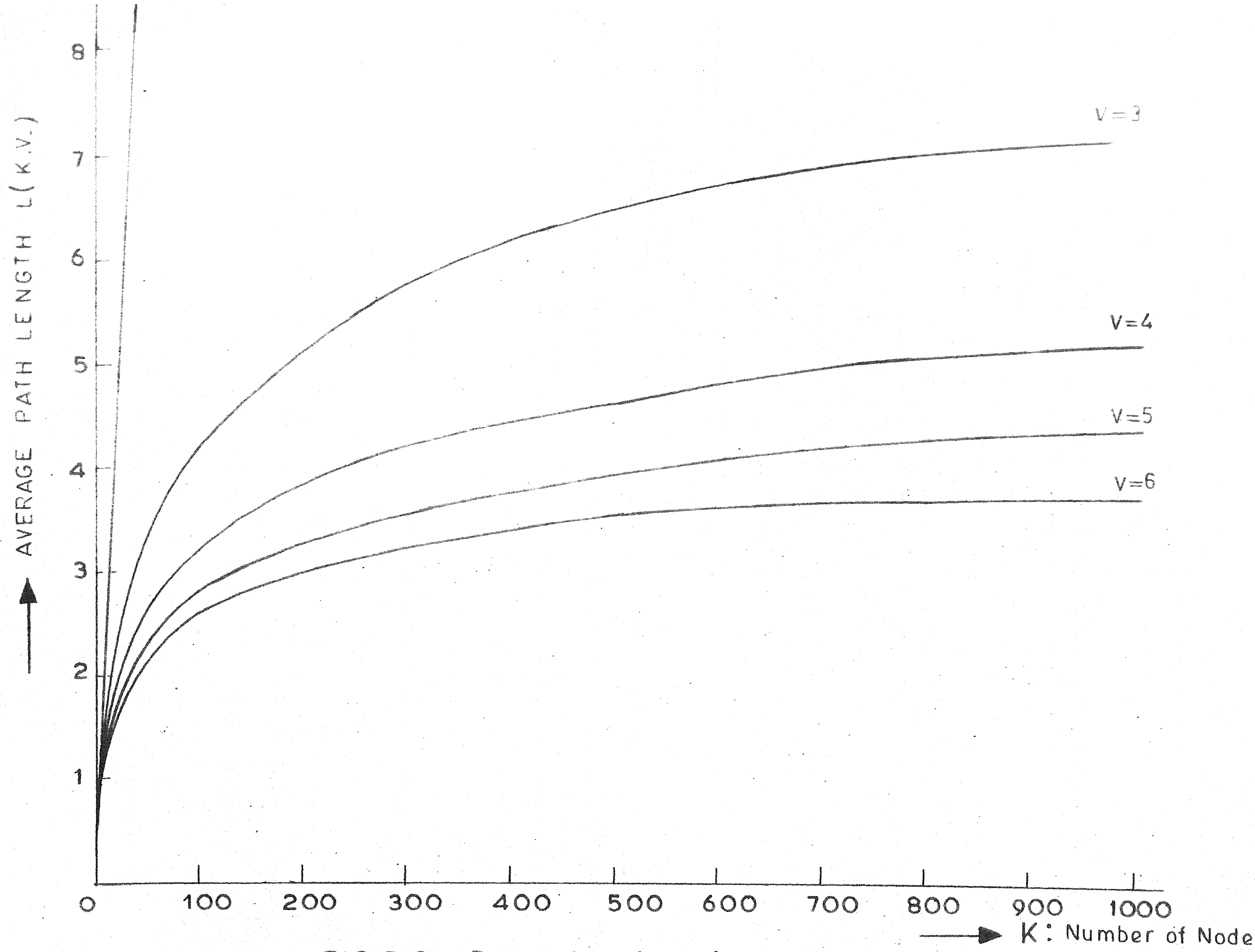


FIG 5.2 Plot of L (K,V)

CHAPTER 6

TRANSMISSION OF PACKETS OVER A CHANNEL

6.1 Introduction:

In the previous chapters while discussing about computer communication networks it is assumed that digital data is carried over a link from one node to the other in the form of packets, without changing any of the characteristics of the data. This is the case of an absolutely ideal system so far as the channel is concerned. In practice no real channel is an ideal one and hence system designs must be carried out so that data is received at a node with high degree of fidelity. Error occuring if any, must be corrected. Besides, depending on the channel parameters considerations have to be given to the data rates, bandwidth requirements and the efficiency of the communication system. In this chapter we will discuss the various channel impairments due to a physical channel and other factors which corrupt the data while being transmitted through a channel, along with various techniques and systems which are needed to combat the effects of the channel.

First of all, for transmission even over a reasonable distance, digital data has to be converted

into an analog form suited for transmission over the channels. This is achieved by a modem which also takes into account the effects of noise. Other signal impairments have been briefly dealt with. Error correcting codes might be required in some cases for correct decisions on received data. Intersymbol interference is another source of error at high data rates. Equalization is needed to counter the intersymbol interference. Thus a system block diagram of a computer communication network with the input/output devices would look like as shown in Figure 6.1. Interface forms part of a computer system. The modems and the communication channel forms part of the associated communication system.

In general the same communication link also carries other kinds of traffic i.e. voice, video etc. and so suitable multiplexers might be incorporated in the system design. At times computer communication network might be a dedicated link. Referring to Figure 6.1 the blocks before the boundary B and after the boundary C form part of the computer system. We will study in this chapter about the communication system when packets cross the boundary B into the system and leave boundary C out of the communication system into the computer system.

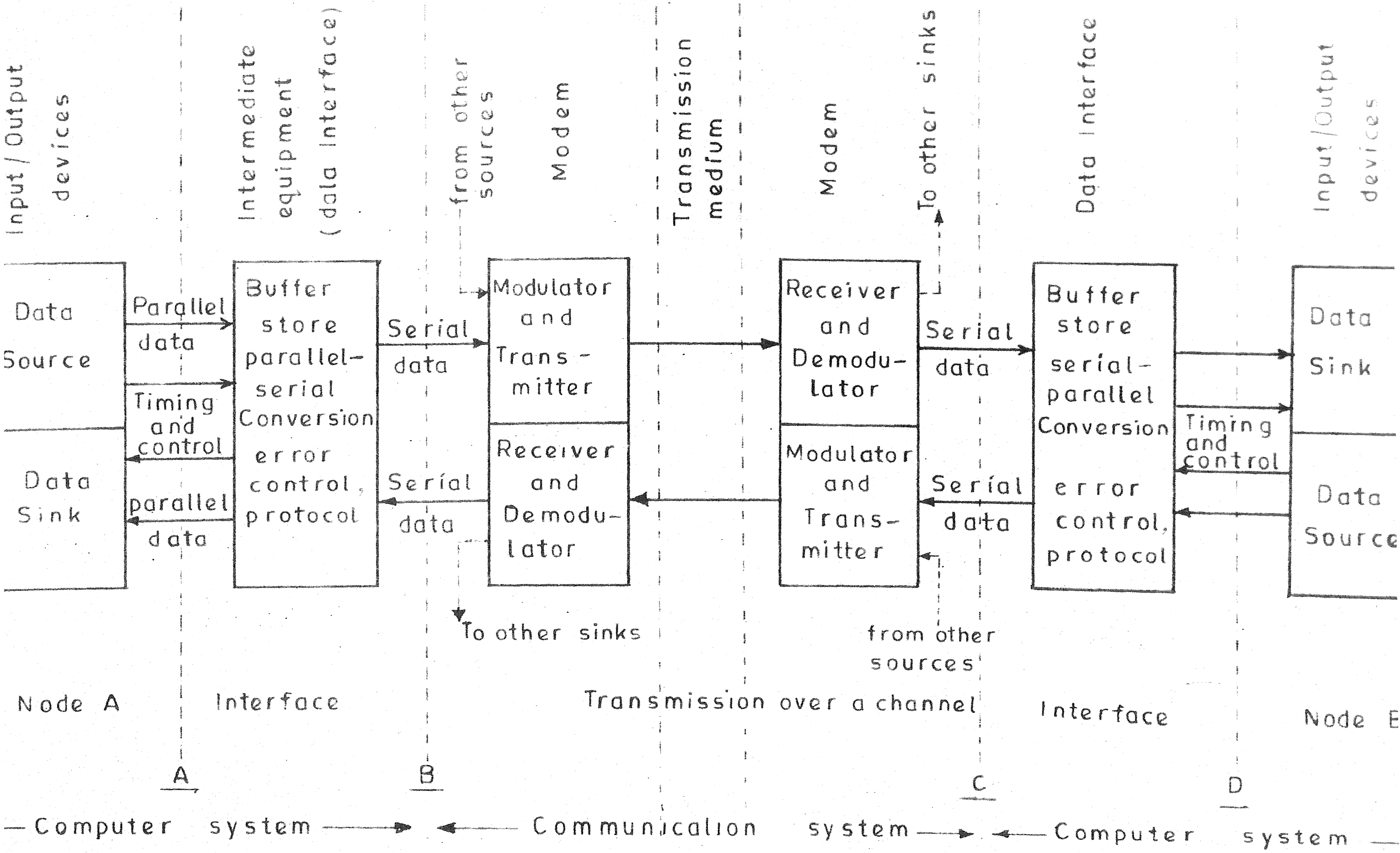


FIG.6.1 Functional block diagram of a computer communication

6.2 Functional Block Diagram:

Figure 6.1 can be viewed as a simplified functional block diagram of a packet switched computer communication network in which the input devices may include keyboard sending units, card readers, paper tape readers and magnetic tape readers etc. The output devices can be printers, magnetic tape recorders, visual displays etc. Data interface is necessary to provide for serial to parallel conversions or parallel to serial conversions as well as other control functions. Timing signals are needed for synchronization. Some type of data conversion equipment is required to change the digital signal to a form suitable for transmission over the medium (analogue channel) and convert it back into the digital form at the receiver end. This function is performed by the modems. Coding is undertaken to alleviate transmission inregularities to enable error detection and corrections and to provide message security. But it is achieved at the cost of reduced bit rate.

6.3 Modulation and Coding:

If one considers the flow of signals in a data communication system, it may be desirable to consider segmented blocks of encoder followed by the modulator at

the transmission end and correspondingly a demodulator and a decoder at the receiver end. Referring to the Figure 7.1, a stream of digital data in the form of packet enters the communication system at B and leaves in the same format into the computer system at C. In between it is modulated in the analogue form suitable for transmission over the medium and then again demodulate in the digital form at the receiver end. In an ideal channel the sequence of digits at the output is the same as that of the input. However, because of noise and other channel impairments the output sequence is seldom a replica of the input sequence. Since error free transmission is of utmost importance in a computer communication system encoding in the form of addition of some redundant bits per packet (say) based on some rule is resorted to so that message digits can be protected from the errors. At the receiver end the message digits are corrected from the total received digits and passed on to the computer system. This process is known as decoding. The communication system of Figure 7.1 can be modified as shown in Fig. 7.2 where channel encoder and channel decoder take care of the errors occurring in the channel. As mentioned above, even though the modulation and coding form part of the same system, for system

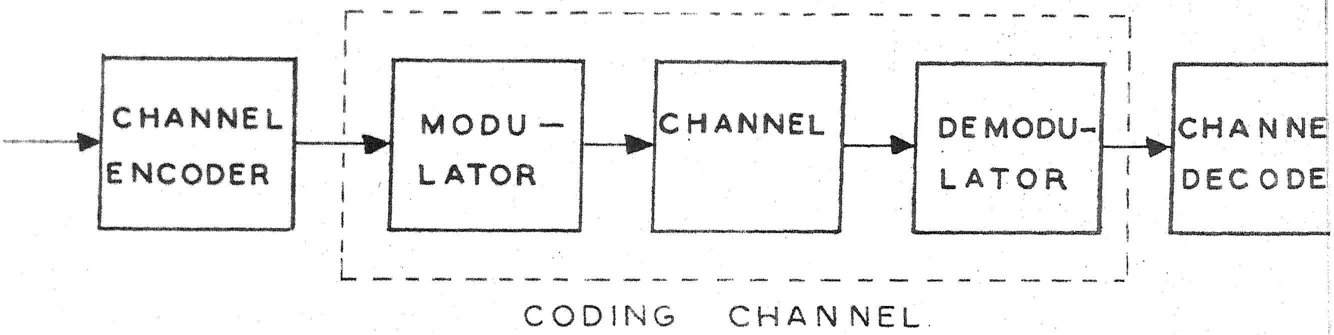


FIG.6.2 Modified block diagram of a digital communication system.

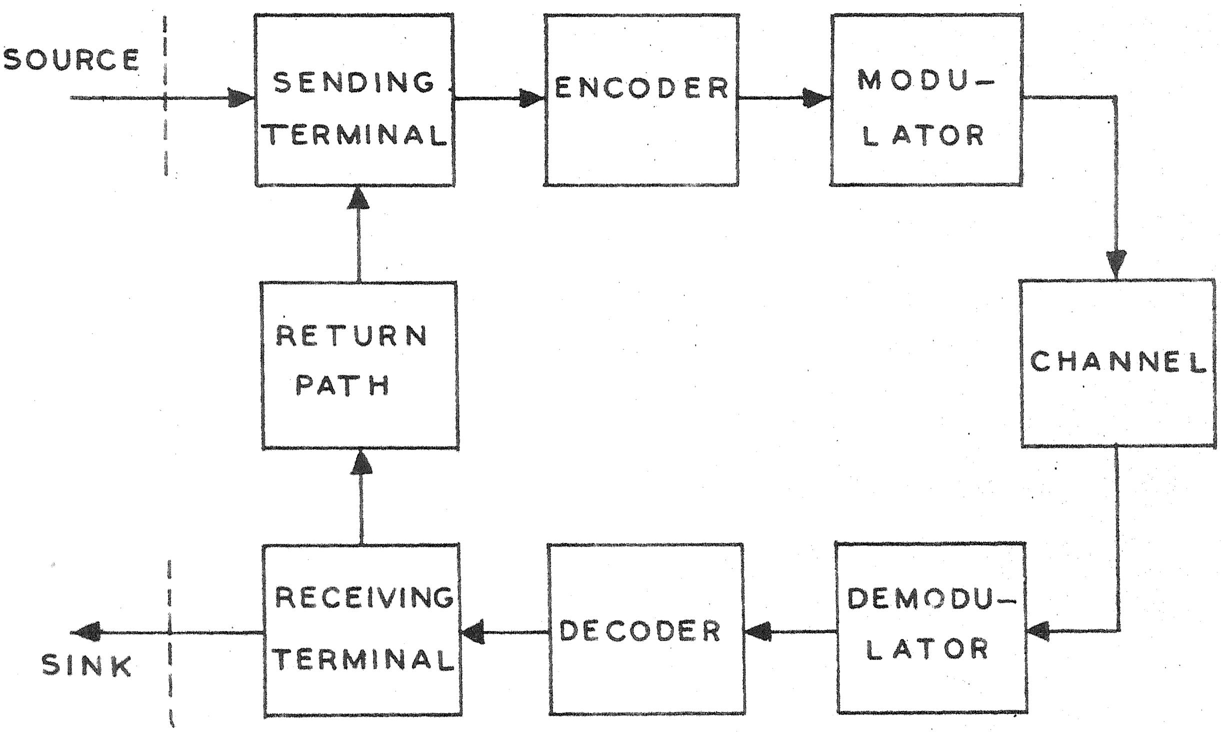


FIG.6.3 ARQ System.

analysis it is appropriate to model them as separate channels. While the modulation channel comprises of the transmission medium and the transducers needed to couple the signal into the medium, the coding channel is a digital link between the encoder and decoder and comprises of the modulator, modulation channel and the demodulators as shown in Fig. 6.2.

The modulation channel covers the modulation and demodulation schemes, some of which are discussed in a later section. The coding channel involves coding decoding schemes and detection or correction of digital errors. Error detection systems are relatively simpler than error correction systems and are called ARQ (automatic repeat request). Here the decoder computes the parity bits from the received data and compares them with the received parity bits. If there are no discrepancies, the packet is delivered to the next node and the sending node is notified through a suitable return channel. If there are any discrepancies then a request for retransmission of the packet is sent by the receiver from the sending node. Alternately, the transmitter waits for an acknowledgement of correct reception of a packet. If the acknowledgement is not received within a specified time, the packet is retransmitted. The block diagram for ARQ system is shown

in Fig. 6.3. With this system the erroneous data is delivered to the sink only if detector fails to detect the presence of errors.

In error correcting technique called the forward error control (FEC) the decoder determines the location of errors from the pattern of discrepancies between the calculated and received parity bits. Decoder, which is a more complicated device, then corrects them. An erroneous data is delivered to the link if the decoder fails to determine the exact location of errors. There is no return path and hence no retransmission of data.

There are various types of codes and coding techniques which are used depending upon the channel characteristics, data rates, and the accepted probability of error.

6.4 Transmission Impairments [2, 20, 21]

Transmission of data over a channel is subject to several types of impairments of data. Some of these are associated with variations in the channel and others are extraneous interferences. All of these result in a distorted replica of the transmitted signal. The impairments limit the data rates and accuracies. The impairments can either be modelled as a deterministic one or a stochastic one

depending on the parameters. For example, deterministic impairments may include:

1. Amplitude and delay distortion
2. Non-linear distortion
3. Frequency offset.

Examples of stochastic impairments would include the effects of:

1. Channel behaviour
2. Phase jitter

Amplitude and delay distortions are the linear distortions affecting a signal during transmission. Delay distortion is caused by the inequality of propagation times at different frequencies within the pass band. It results from the capacitive and inductive reactances of filters used in the communication system. Such variations limit the speed of transmission and reduce the margin of error. Delay distortion becomes more critical as data speeds increase. Typically, higher data speeds are achieved by shortening the width of each pulse. Because of the shorter pulse width, slight shift in relative phase between frequency components have a greater effect in distorting the signal. The successive pulses interfere with one another at sampling times. This intersymbol interferences increases the error rate.

Amplitude distortion is caused by variation of transmission loss with frequency. The resultant effect is a fairly definite limitation on the bandwidth and rate of signalling.

Non-linear distortion result from many sources such as saturation effect in magnetic cores, non-linear characteristics of circuitary, and crowding in multi-channel modulators and amplifiers. These non-linearities cause harmonic distortions and the creation of spurious frequencies. Nonlinear distortion causes intersymbol interference just like linear distortion but generally of a lower magnitude.

Frequency offset is a small shift of all the transmitted frequencies due to the small difference in modulating and demodulating frequencies in the carrier system. This type of impairment is usually not serious in data channels except for those using narrow band frequency modulation.

White Gaussian noise caused by the thermal agitation of electrons is always present in electrical circuits and can not be eliminated. It has a fairly uniform power spectrum across the band. Additive Gaussian noise

is the most commonly used model of a noisy channel as it is mathematically tractable. At low bit rates the effect of Gaussian noise can be easily combated by suitable increase in signal power and hence it does not introduce a very serious problem.

Impulse noise is a primary source of errors in the transmission of data. It is sporadic and occurs in brusts on discrete impulses called hits. Impulse noise is extremely unpredictable and is commonly caused by electrical storms and the operation of switching equipment in the system. This is very common if we use telephone network for our purposes.

Inaccuracy regarding the instants of phase change at high data rates results in phase jitters and might become a serious impedament if MPSK modulations techniques (e.g. 8 PSK or so) are used in modems.

6.5 Baseband Signalling:

The job of the transmitter is basically of that of frequency translation and power amplification which are needed for transmission of signals over the channel. Most efficient frequency translation systems are linear and hence the effect of linear translation can be considered

at the baseband channel. That is, all the signal processing is done at the baseband level where signals upto the d.c. frequency level are considered. Also, baseband signalling caters for flow of the signals in local loop communications like from terminal to a computer. The study of baseband signalling is relevent because many of its concepts and parameters carry over directly to modulation (linear translation). Further, the performance characteristics of baseband transmission serve as useful standards when comparing the various types of modulation. Thus, we are interested in transmission of a sequence of digits (a_0, a_1, a_2 ....) each $a_i \in (0,1)$ over the channel. This stream of binary data might be transmitted by any suitable waveform (basic pulse shape). Therefore, mathematically the sequence can be represented as [2, pp. 157]

$$s(t) = \sum_n a_n g(t-nT) \qquad (6.5.1)$$

where a_n is the amplitude of nth pulse depicting whether it is a0 or a1.

g(t) is the basic pulse shape

T is the pulse duration

If the channel is linear and distortionless over all frequencies (has infinite bandwidth) g(t) suffers no

degradation in transmission and a large signalling rate can be achieved. But a real channel has a finite bandwidth and the frequency response is not ideal. This causes spreading and overlaping of pulse and results in intersymbol interference. Conventionally the stream of data bits are reconstructed at the receiver end by sampling and deciding whether a0 was transmitted or a1 during that in pulse duration.

The received waveform y(t) which is the sum of the transmitted signal and the noise is a sampled at the kth instant, sampling time kT, and can be written as

$$y_k = y(kT) = x_o\left(a_k + \frac{1}{x_o} \sum_{\substack{n \\ n\neq k}} a_n x_{k-n} + \frac{\eta_k}{x_o}\right) \tag{6.5.2}$$

where $x_{k-n} = x(kT-nT)$ and x(t) is the shape of the output pulses.

In equation (7.5.2) the factor x_o represents the gain (or attenuation) of the signal in passing through the system. The first term represent the desired output level. The second and the third term represent the intersymbol interference and noise respectively. Therefore the aim of designing a baseband system for reliable data communication

is to minimise the ISI and noise. Further, we should maximise the signalling rate for a given bandwidth.

It is known (18) that raised cosine waveshapes will not permit any ISI if sampled at the Nyquist rate and hence te the pulse shape and correspondingly the transmitter filter is to be so designed that raised cosine characteristics are achieved. Thus, the corresponding bandwidth requirement is at least $\frac{1}{2T}$ Hz (signalling rate $\leq$ 2B).

Since the amplitude of a pulse is zero at the sampling instants of other pulses it does not affect the amplitude of other pulse and hence no ISI.

Between extremes of Nyquist pulses ($B = \frac{1}{2T}$) and rectangular pulses there is a compromise made possible to eliminate ISI by folding operation. In the usual case the pulse spectrum x(w) occupies less than twice the minimum of π/T Hz of bandwidth, and has raised cosine characteristics.

6.6 Modulation Techniques:

Digital data as used in a computer is represented by square waves. However such waveforms have to be suitably modified before frequency translation and transmission over a link so that efficient transmission may be

achieved. Thus these pulses are made to modulate the carrier to vary the carrier's amplitude, frequency, phase or some combination of these.

Double sideband AM was used in the earliest binary data transmission but has now been replaced by FSK. FSK is the usual choice where simplicity and economy are more important than bandwidth efficiency. Today amplitude modulation is used in conjunction with PSK to achieve high data rates in high speed modems. In some cases AM is used for multilevel transmission.

PSK modulation techniques are being increasingly used in the last decade or so because these are relatively insensitive to transmission impairments. In binary phase shift keying (BPSK) the carrier phase is shifted by 0 or 180 degrees depending on whether a one is transmitted or a zero. The corresponding detection requires a precise phase reference. Differential PSK, is used to get over the synchronization problems. Here the signal from the previous bit interval is used as a phase reference for the current bit interval. Since the phase reference signal is not smoothed over many bit intervals the performance of DPSK is somewhat inferior to PSK. The errors occur in pairs. Quaternary PSK (QPSK) involves

encoding two bits at a time into one of the four possible carrier phases spaced 90 degrees apart. Like PSK, data in QPSK can be differentially detected. Most of the form phase modems use phase shifts of $\pm 45°$ and $\pm 135°$ or $0° \pm 90°$ and $180°$ as shown in Fig. 6.4.

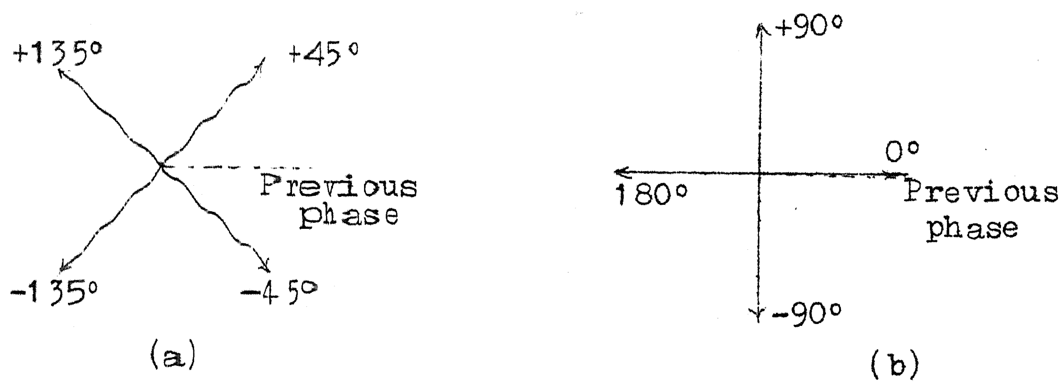


Fig. 6.4 Choices of Phase Shift for QPSK

The former has the advantage of continuous phase changes which aids in symbol synchronism.

The ever increasing need for bandwidth conservation and high data rates has led to the use of a class of hybrid AM/PM techniques called Quadrature amplitude modulation (QAM). In a 4 Level QAM twelve values of phase shifts, and 3 values of amplitude give form bits per symbol.

The standard high speed rates for voice band modems are 4800, 7200 and 9600 bits per sec. Upto 2400 bits

per sec. it is termed as low speeds. Modems handle very high data rates by accepting and delivering binary signals over parallel interface leads which are carried over the same voice channel. For data speeds upto 2400 bits per sec. FSK is the clear choice. For 2400 bits per sec. DPSK has become the worldwide standard.

From 4800 bits per sec to 9600 bits per sec there is a wide choice of modems as shown in Table 1. The maximum symbol rate is limited by the available bandwidth while the number of symbol starts depends to a great extent on nonlinear distortion and phase jitter.

Table 1

Types of Modulation and Bandwidth Requirements for high speed modems

Type of modulation	Bit/ cycle	Nyquist band required for 4800	7200	9600 bits per se
2 level USB or QAM	2	2400		
4 Ø PSK	2	2400		
3 level QAM	3	1600	2400	
4Ø + 2AM levels	3	1600	2400	
8Ø PSK	3	1600	2400	
4 level QAM	4		1800	2400
8Ø + 2 AM levels	4		1800	2400
6 level QAM	5			1920
8 level QAM	6			1600

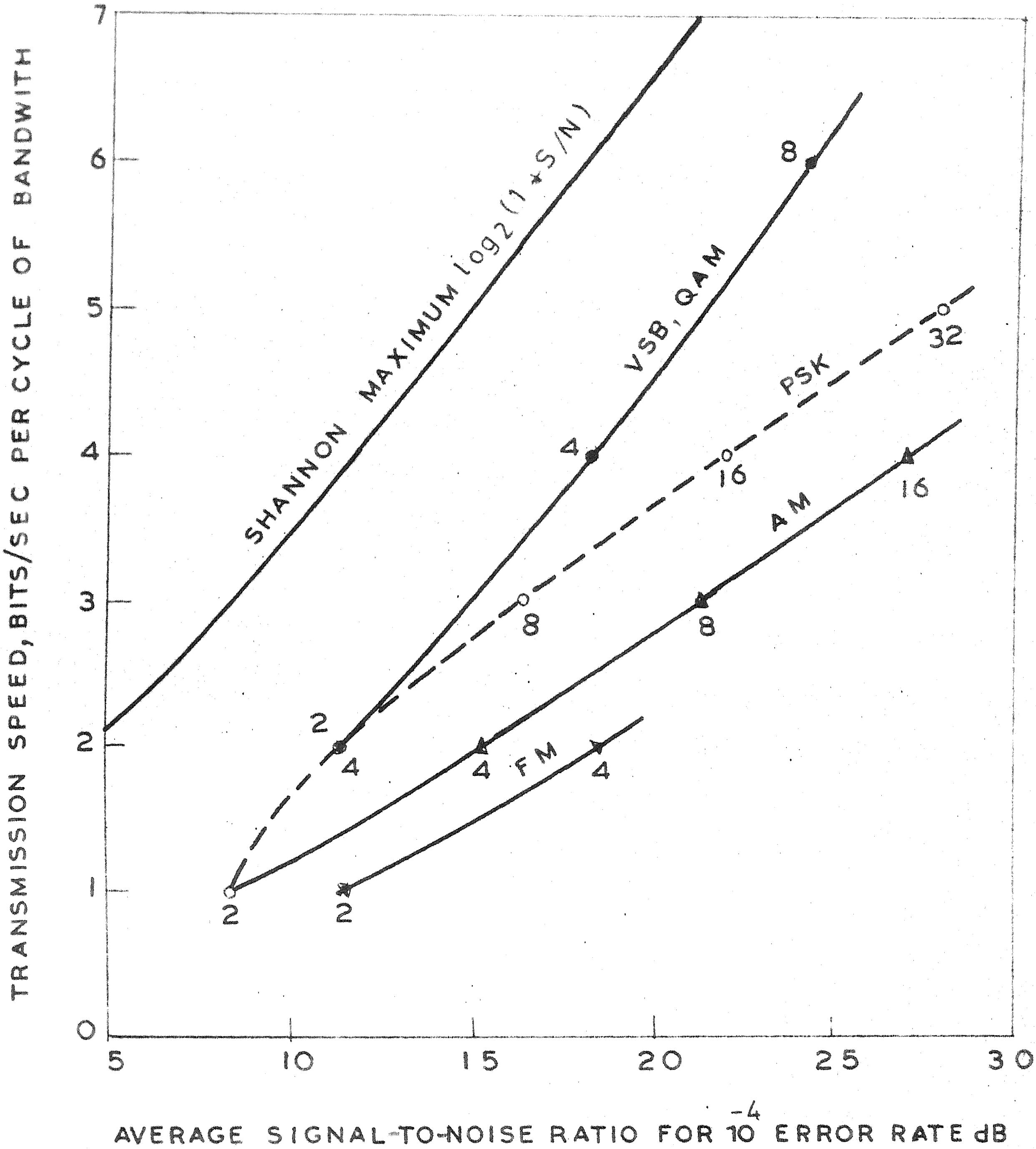


FIG.6.5 Comparison of modulation methods.

Comparison of the performance of modems using different modulation schemes is provided in Fig. 6.5.

6.7 Equalization:

Delay distortion becomes more critical as data speeds increase. Typically, higher data speeds are achieved by shortening the width of each pulse. Therefore slight shifts in relative phase between frequency components have greater effect in distorting the signal. Intersymbol interference which is defined as the distortion caused by the tails of preceeding and succeeding pulses intruding into the time slot of the pulse currently being transmitted, is of great concern in high speed data transmission. Intersymbol interference results from the non-uniform delay and amplitude characteristics of communication channels. The raised cosine waveshape may not always be feasible to implement. An equaliser - basically an adjustable jitter is used to introduce a controlled amount of delay at certain frequencies to achieve uniform delay and amplitude variation over the entire bandwidth. At lower transmission speeds say upto 2400 bits per sec - it is usually adequate to use a fixed compromise equaliser chosen to match the mean of the range of expected delay distortion. In other words, average performance of the channel is

determined by taking a statistical sampling of signals and then the equalizeris designed to compensate for average distortion characteristics. However, for higher speeds more efficient utilisation of bandwidth is required, and hence equalization is needed which is more precisely tailored to the particular channel in use during any one random connection. In a situation where channel characteristics vary randomly automatic equalisers are used which usually is in the form of a transversal filter - a tapped dela line with associated gains. By varying the gains of the taps, the controlled amplitude and delay distortion introduced into the signal components can be set to compensate for the effects of the channel. A typical adaptive equalizer is depicted in Fig. 6.6.

Adaptive equalization provides for constant monitoring of the data signal and adjustment of the tap gain settings to give optimum equalization at all times. In adaptive equalization data transmission is preceeded by a short preamble (fixed packet head) during which test pulses allow determination of an initial equalization level and setting of tap gains accordingly. As data are

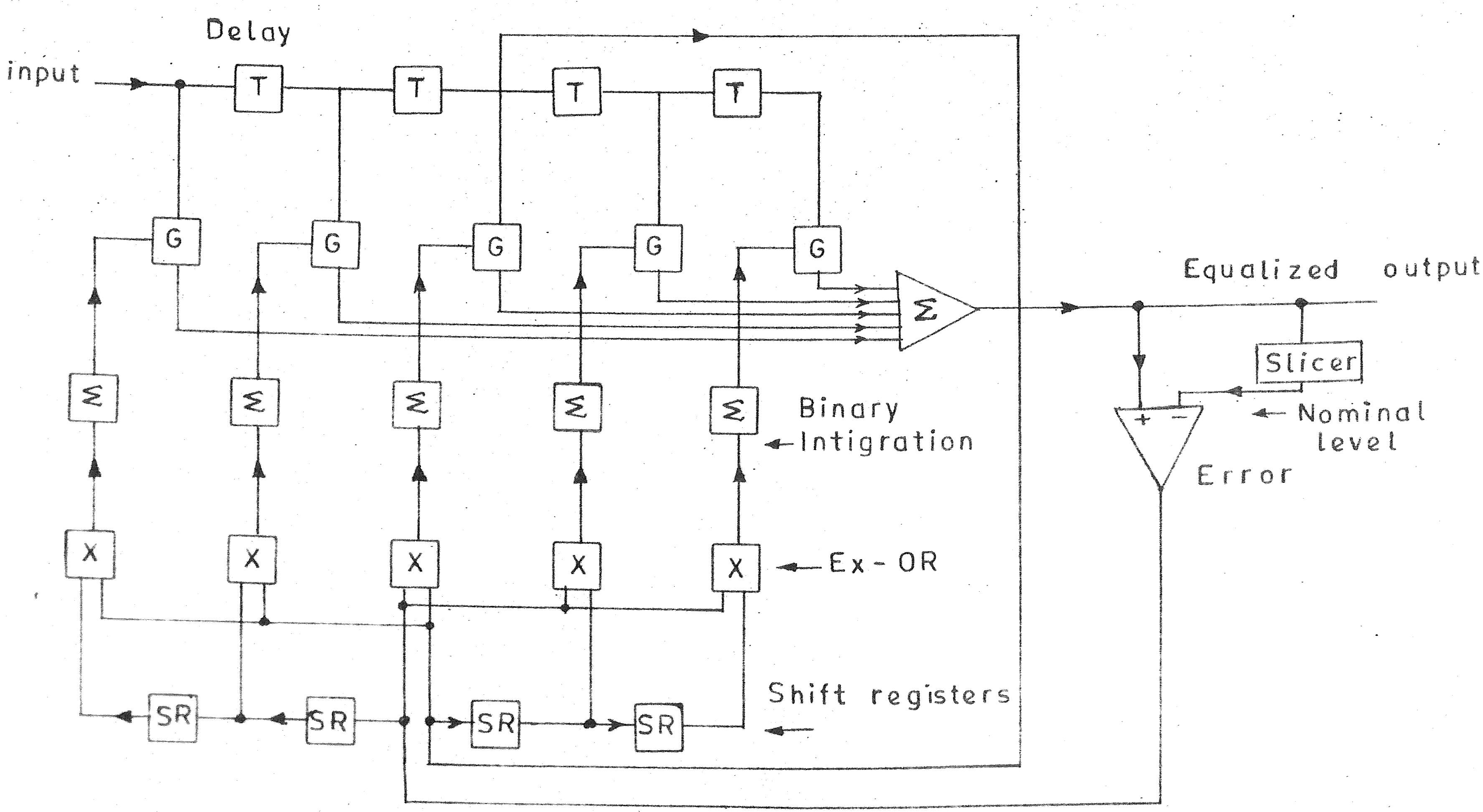


FIG. 6.6 Adaptive Equalizer

transmitted samples are taken and compared to the equalization level. Any difference between data sample and equalized sample results in the generation of an error sample and the setting up of tap gains to cancel the difference.

CHAPTER 7

A TYPICAL COMPUTER COMMUNICATION SYSTEM

We have discussed various aspects of a computer communication system in the preceeding chapters. Here we undertake to exemplify a particular system which might be suitable for defence applications. The parameters of such applications are high reliability and low probability of BER.

A typical physical modelis as shown in Fig. 7.1 which can be modified in a systematic form as in Fig. 7.2

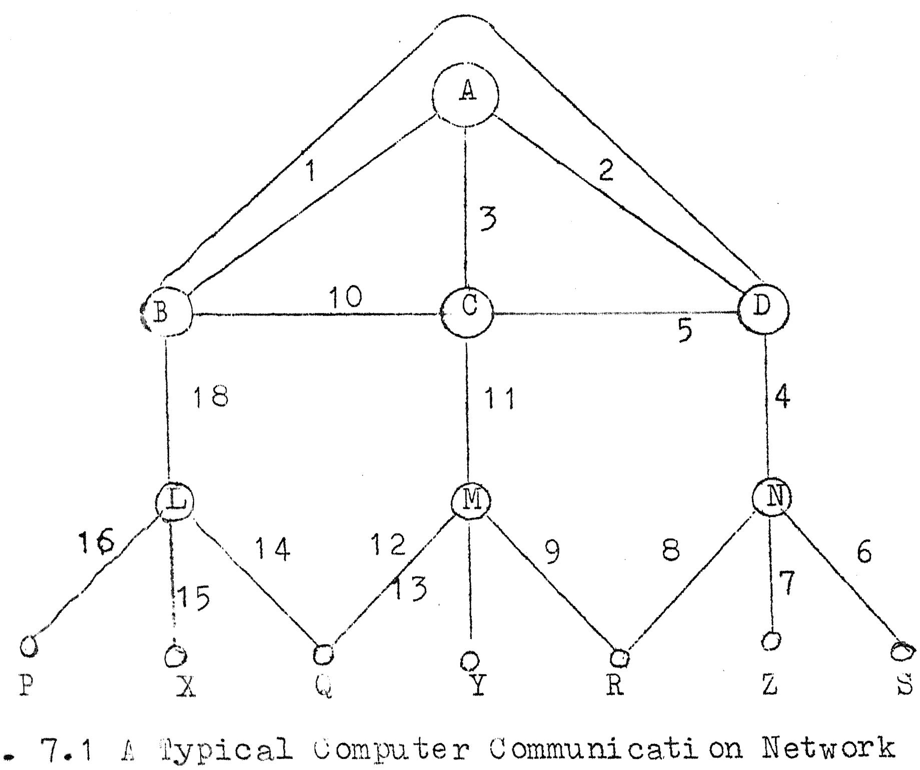


Fig. 7.1 A Typical Computer Communication Network

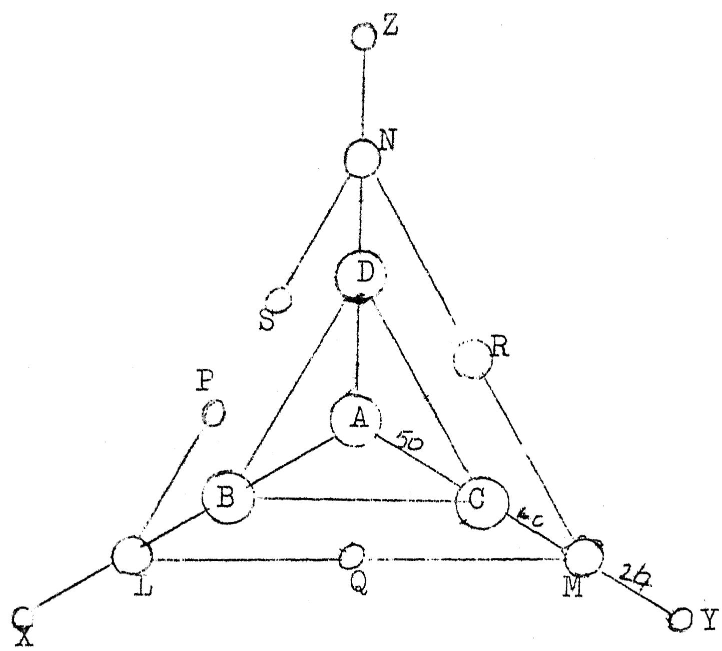


Fig. 7.2 Modified Computer Communication Network

There is a very large central computer (A) at the main centre which interacts with large computers (B,C,D) at sub-centres. These smaller computers are interlinked amongst themselves as well as are connected to the auxilliary computing facilities at L,M,N...etc. Some of these are also interconnected. The main centre and the sub-centres are connected with each other by full duplex high speed communication lines whereas the others can be

connected with medium speed lines. The development of the network involves following principal activities;

1. The design of the IMPs, concentrators and multiplexers to act as nodal store and forward switches.

2. The design of protocol to meet the users requirements.

3. Topological modelling and design to specify the capacity and location of each communication circuit within the network.

4. System modelling and measurement of network performance.

Having specified the needs, e.g. routing, flow control, error control and the user protocols etc., standard designs of IMPs and protocols are available. Of great importance is the property that all IMPs operate with identical programs and the network should accomodate the expansion of the number of IMPs without changing the overall performance. This covers the first two factors for which only brief mentions have been made in this thesis. We have gone into the details of the topological design considerations and the results are mentioned here for the network of Fig. 7.1.

We know the geographical locations of the computers and terminals. Our other requirements are:

1. Very high reliability
2. Minimum message delays
3. Minimum path length
4. Higher capacity for main centre and sub-centre channels, than the auxilliary channels.
5. Routing techniques and protocols for efficient transmission of packets and to cater for the message protocols of the services.
6. To avoid congestion at various nodes specifically for the nodes A,B,C,D.
7. Error control to achieve BER of 10^{-6}.
8. Modems to suit the existing channels.

Network performance has been measured for various criterion between some node pairs only. This will indicate to us the performance between the other pairs of nodes though it can be exactly calculated.

We calculate the network performance with the following parameters.

1. Packet length = 1000 bits
2. Average message length = 5000 bits
3. Capacity as shown in Fig. 7.2

Table I: Message Delays

Traffic intensity →

Node pair ↓	.1	.15	.2	.25	.3	.35	.4	.45	.5	.6	.7
(AB) (AC) (AD) (BC) (BD) (CD)	.1223	.1303	.1384	.1481	.1596	.1734	.1882	.2103	.2324	.3045	.444
(AL, AM, AN)	.1772	.1914	.2056	.2251	.2468	.2782	.3114	.3735	.4353	.7577	3.635
(BL, CM, DN)	.1574	.1715	.1853	.2051	.2266	.2581	.2918	.3533	.4151	.7374	3.615
(AP, AX, AQ, AY, AR, AZ, AS)	.2183	.2323	.2469	.2662	.2875	.3192	.3527	.4144	.4765	.7981	-
(BP, BX, BQ, CQ, CY, CR, DR, DZ, DS)	.1982	.2124	.2266	.2461	.2678	.2992	.3324	.3945	.4563	.7787	3.656

Table II

Reliability of individual links has been assumed as follows.

Link	Reliability
1	.98
2	.98
3	.98
4	.90
5	.95
6	.90
7	.90
8	.90
9	.90
10	.95
11	.90
12	.90
13	.90
14	.90
15	.90
16	.90
17	.95
18	.9

Based on the above reliabilities of the individual links we have calculated the system reliability between various nodes as shown in Table III. The system reliability calculations are as per the method discussed in Chapter 4.

Table III

The reliability calculations for various node pairs

Node pair	System reliability
(AB)	.99997
(BM)	.98388
(AP)	.89305
(AY)	.89301
(BS)	.88521
(BR)	.98225
(BQ)	.98225
(AM)	.99222

We can get the idea of the system reliability between other node pairs from the above table.

The exact average path length between the node A and all others is difficult to calculate. However we can get an intitutive picture from the method discussed in Chapter 5. We see that the average path length for 14 nodes with valence 4 is 1.89 and that for 14 nodes with valence 3 is 2.23. Since the network discussed in this chapter comes very close to a network with 14 vertices and valence four we can put our average path length to be approximately 1.9.

Discussing briefly about the communication aspect, we have taken the data rate of modems between the main centre and sub-centres and between the sub-centres as 9.6 K bps. Between the others, modems of 4.8 K bps can be used. Margin has been given for an increased number of nodes. For 9.6 K bps, modems using QPCK are suggested and for 4.8 K bps modems using BPCK are suggested. The SNR requirements then would be for Pe of 10^{-6} 12 and 10.5 dB respectively.

CONCLUSION

In the preceeding chapters, some aspects of computer communication systems have been explained. The first is the need for packet switching. Circuit switching techniques applied in telephone networks are not suited for computer communication because of the various reasons mentioned in Chapter 2. It has been possible to look at the overview of packets and packet switching, and not go into the design aspects for its implementation. Nevertheless we get the idea of the organisation of the system.

Message delays, reliability and the average path length are the important factors affecting the topological designs of a computer communication network and have been described in the third, fourth and the fifth chapters respectively. Since network design is a complex task it is important to provide the designer with simple criteria for evaluation of the system.

In Chapter 3 the distribution of the overall message waiting and delay times, over an N channel path in a packet switched communication network have been described. The messages have been assumed to be of random length with random arrival. They are divided into fixed length

packets for independant transmission. The overall message waiting time distribution follows the waiting time distribution derived for an M/G/1 queueing system. Time delays for exponentially distributed message lengths have been plotted in Fig. 3.4 for various traffic intensities. Time delay problems for more involved topological and flow situations are currently under study. The problem of message time delay calculation when the communication path is composed of K vertex disjoint paths is equivalent to the queueing system with K servers.

Graph theoretic results have been found to be very useful in characterizing computer network reliability. The probabilistic reliability criterian analysis are more difficult than deterministic reliability criterian. Various reliability measures have been listed in chapter 4 and the one based on minimal cuts has been described. It gives a good lower bound to the system reliability and reduces computational problem.

The average path length explained in Chapter 5 provides a lower bound for graphs with K nodes and valence V. It is one of the methods suggested for particular type of networks since it provides a convenient type of model for reliability analysis based on connectivity. The relationship

between valence and average path length allows an examination of the trade off between the transmission delay and number of transmission lines in a computer communication system.

The existing transmission links were not designed for transmission of computer data. In telephone networks, all its design features from the terminal equipment to the switching and transmission are determined by speech and conversational requirements. Attempt has been made in Chapter 6 to explain briefly how the existing transmission links are used in the computer communication. systems.

What we have seen are the few aspects of computer communication systems using packet switching, involving computer networks and communication systems. Literature on the various aspects of the system conceived with the amalmagation of computer and communication technologies is scattered in bits and pieces. It is extremely desirable to bring together as many aspects of it as possible for the need of technical community. Also we have not carried out the protocol design and routing algorithm. Any packet switching network must have a fixed set of protocols and routing arrangements. These need be discussed for a comprehensive picture of computer communication system.

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